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*Technology
Assessment*

The Intelligibility of Amplitude Companded Sideband Modulation Compared to Conventional Frequency Modulation

NIJ Report 201-87

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The Technology Assessment Program is an applied research effort that determines the technological needs of justice system agencies, sets minimum performance standards for specific devices, tests commercially available equipment against those standards, and disseminates the standards and the test results to criminal justice agencies nationwide and internationally.

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James K. Stewart, Director
National Institute of Justice

U.S. Department of Justice
National Institute of Justice

Technology Assessment Program

The Intelligibility of Amplitude Companded Sideband Modulation Compared to Conventional Frequency Modulation

NIJ Report 201-87

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June 1989

**U.S. DEPARTMENT OF JUSTICE
National Institute of Justice**

James K. Stewart, Director

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FOREWORD

The Law Enforcement Standards Laboratory (LESL) of the National Bureau of Standards furnishes technical support to the National Institute of Justice (NIJ) program to strengthen law enforcement and criminal justice in the United States. LESL's function is to conduct research that will assist law enforcement and criminal justice agencies in the selection and procurement of quality equipment.

LESL is: 1) Subjecting existing equipment to laboratory testing and evaluation and 2) conducting research leading to the development of several series of documents, including national voluntary equipment standards, user guides, and technical reports.

This document covers research on law enforcement equipment conducted by LESL under the sponsorship of NIJ. Additional reports as well as other documents are being issued under the LESL program in the areas of protective equipment, communications equipment, security systems, weapons, emergency equipment, investigative aids, vehicles, and clothing.

Technical comments and suggestions concerning this report are invited from all interested parties. They may be addressed to the Law Enforcement Standards Laboratory, National Bureau of Standards, Gaithersburg, MD 20899.

Lester D. Shubin
Program Manager for Standards
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COMMONLY USED SYMBOLS AND ABBREVIATIONS

A	ampere	H	henry	nm	nanometer
ac	alternating current	h	hour	No.	number
AM	amplitude modulation	hf	high frequency	o.d.	outside diameter
cd	candela	Hz	hertz (c/s)	Ω	ohm
cm	centimeter	i.d.	inside diameter	p.	page
CP	chemically pure	in	inch	Pa	pascal
c/s	cycle per second	ir	infrared	pe	probable error
d	day	J	joule	pp.	pages
dB	decibel	L	lambert	ppm	part per million
dc	direct current	L	liter	qt	quart
°C	degree Celsius	lb	pound	rad	radian
°F	degree Fahrenheit	lbf	pound-force	rf	radio frequency
diam	diameter	lbf-in	pound-force inch	rh	relative humidity
emf	electromotive force	lm	lumen	s	second
eq	equation	ln	logarithm (natural)	SD	standard deviation
F	farad	log	logarithm (common)	sec.	section
fc	footcandle	M	molar	SWR	standing wave ratio
fig.	figure	m	meter	uhf	ultrahigh frequency
FM	frequency modulation	min	minute	uv	ultraviolet
ft	foot	mm	millimeter	V	volt
ft/s	foot per second	mph	mile per hour	vhf	very high frequency
g	acceleration	m/s	meter per second	W	watt
g	gram	N	newton	λ	wavelength
gr	grain	N-m	newton meter	wt	weight

area = unit² (e.g., ft², in², etc.); volume = unit³ (e.g., ft³, m³, etc.)

PREFIXES

d	deci (10 ⁻¹)	da	deka (10)
c	centi (10 ⁻²)	h	hecto (10 ²)
m	milli (10 ⁻³)	k	kilo (10 ³)
μ	micro (10 ⁻⁶)	M	mega (10 ⁶)
n	nano (10 ⁻⁹)	G	giga (10 ⁹)
p	pico (10 ⁻¹²)	T	tera (10 ¹²)

COMMON CONVERSIONS

(See ASTM E380)

ft/s × 0.3048000 = m/s	lb × 0.4535924 = kg
ft × 0.3048 = m	lbf × 4.448222 = N
ft-lbf × 1.355818 = J	lbf/ft × 14.59390 = N/m
gr × 0.06479891 = g	lbf-in × 0.1129848 = N-m
in × 2.54 = cm	lbf/in ² × 6894.757 = Pa
kWh × 3 600 000 = J	mph × 1.609344 = km/h
	qt × 0.9463529 = L

$$\text{Temperature: } (T_F - 32) \times 5/9 = T_C$$

$$\text{Temperature: } (T_C \times 9/5) + 32 = T_F$$

LETTER FROM THE DIRECTOR

Technology can extend the impact of law enforcement. If technological innovations are faulty or poorly designed, however, they can severely handicap us and cost lives. Through its Technology Assessment Program (TAP), the National Institute of Justice provides reports like this one, based on careful, scientifically designed tests, to help agencies large and small apply new and emerging technologies.

Police chiefs and associations have pressed for ways to alleviate the current shortage of communications channels, a critical problem for many police departments. Larger cities and several others near our Canadian and Mexican borders simply do not have enough frequency spectrum to accommodate their communication needs. They must "make do" with the channels presently allocated to them by the Federal Communications Commission.

Open space on the radio frequency spectrum is becoming a scarce commodity. Law enforcement has access to only four general bands, where congestion may eventually cause gridlock. A technological breakthrough on this problem could vastly improve the use of law enforcement frequencies. To guide efforts to relieve frequency spectrum congestion, the Communications Committee of the Technology Assessment Program Advisory Council recommended research to evaluate and critique new technologies.

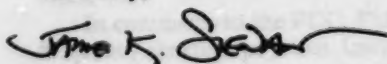
Assuming that the public cannot acquire additional bands, we must be more efficient in using the ones we have. A new technique to narrow existing channels promises to squeeze five channels into the space now used for one and maintain quality voice transmission with enhanced privacy. This technique, called Amplitude Companded Sideband (ACSB), contains potential for improving spectrum efficiency — if it can be successfully implemented by law enforcement agencies.

To help determine the suitability of ACSB technology for law enforcement use, the National Institute's Technology Assessment Program commissioned the study described in this report. Communications planners and others concerned with this issue are urged to read the report in its entirety. Those who wish only a brief overview of ACSB's suitability for police use may find the executive summary meets their needs.

It appears that implementation of ACSB may be one solution to the growing communication channel shortage, but until it is placed in service its overall effectiveness will not be known. The National Institute of Justice will follow the law enforcement community's use of ACSB systems and, when appropriate, report on users' experience with the technology.

In the meantime, this report, which outlines the pros and cons of the current ACSB technology, will help law enforcement administrators and planners as they tackle present and long-range needs in communication systems. The TAP has done the initial assessment and now recommends real-world tests for more reliable, improved answers.

Sincerely,



James K. Stewart
Director

EXECUTIVE SUMMARY

The Federal Communications Commission (FCC) has authorized the use of a new modulation technique, Amplitude Companded Sideband (ACSB), in portions of the 150-174 MHz Private Land Mobile frequency band. Because ACSB potentially permits as many as six separate channels in the same bandwidth as a single FM communication channel, the potential benefit to law enforcement agencies needing additional public-safety communications channels is obvious.

This report provides historical background on land mobile communications and a discussion of the FCC rulemaking on ACSB for those not familiar with this modulation technique. The functional characteristics of such systems are fully described together with performance measurements from other sources (Stanford Research Institute) and those used in this study. The experience of three operational systems is discussed, and the performance of ACSB systems is compared with that of FM communication systems to enable independent consideration of the applicability of ACSB to law enforcement communication systems.

During the course of this research, a performance standard was also developed, which will be promulgated by the National Institute of Justice for use by law enforcement agencies and other interested parties in procuring and applying ACSB systems to their needs.

At first glance, ACSB sounds like the answer to the prayers of all who are short of frequency spectrum. At a channel spacing of 5 kHz, the use of ACSB will allow four to six channels to be placed where, at present, only one FM channel can be used. But several manufacturers of FM equipment and certain users, perhaps influenced by these manufacturers, tell us that ACSB cannot be expected to perform as advertised. They say that the expected numbers of channels cannot be squeezed into the space provided, and, if they could, ACSB still would not provide the reliability and voice quality we have come to expect from FM systems. They warn about the cost of the new ACSB equipment and tell us to expect maintenance problems as the equipment is subjected to the rugged law enforcement environment.

So law enforcement users are faced with a dilemma. Should they take a chance on ACSB to obtain an urgently needed capability and perhaps suffer the consequences of a degraded system that might not do the job and may even have to be replaced? This experience could prove to be costly. Or should agencies continue with FM, knowing that the system, at best, is not able to fulfill basic needs (due to insufficient channels) but provides a known operational capability?

FM obviously has some advantages over ACSB. It is a known quantity, with proven equipment of acceptable voice quality. The manufacturers tend to be a well-established group with repair and spare parts capability and should be able to support their systems throughout the normal life cycle of use. FM equipment is available in all four authorized police frequency bands.

ACSB, on the contrary, is presently available only in one frequency band. Furthermore, the manufacturers have yet to demonstrate the staying power necessary to support systems in the field throughout their life cycle.

Besides the need for less frequency spectrum, ACSB has at least one other advantage over FM, at least temporarily. Its voice privacy capability has led several companies who need such a capability to use ACSB.

Based on tests conducted for this study and prior results, there seems to be no substantial difference in coverage and range between FM and ACSB, assuming equivalent transmission and reception equipment. ACSB voice quality seems more variable at this stage of development (sometimes better, sometimes worse than FM). As ACSB manufacturers refine their product, the variance should be reduced.

As ACSB has been subjected to greater scrutiny, most proponents are pulling back from proposals for maximum spectrum utilization, i.e., five 5-kHz channels placed in a 25-kHz channel spacing. Instead, they suggest something like three channels be allowed, for a spectrum savings of 3:1 in lieu of the previous 5:1 claims. This approach probably will reduce or eliminate adjacent-channel problems experienced by earlier users.

In comments to the FCC, FM manufacturers have tended to downgrade the ability of ACSB to solve the frequency shortage problem. Instead, they have called for additional UHF TV spectrum to be made available to the public safety community. Public broadcasters, meanwhile, want to hold on to their already-allocated UHF TV spectrum, and they have been advocating the use of ACSB equipment by public safety agencies to make better use of available spectrum. The FCC agrees and has made the frequencies available, but, at present, no U.S. public safety agencies in the United States have taken advantage of this opportunity. Only firms using business radio, another segment of the land mobile industry, have purchased ACSB systems as suggested by the FCC.

In summary, ACSB may be the solution to the growing frequency spectrum shortage. Until it is put into operation by the law enforcement community, however, its real ability to solve the problem will not be known.

A COMPARISON OF AMPLITUDE COMPANDED SIDE BAND MODULATION WITH CONVENTIONAL FREQUENCY MODULATION

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Amplitude companded sideband (ACSB) is a narrowband modulation technique that is now being used in land mobile radio. Land mobile communication is a necessary element of law enforcement activities, and it must meet certain requirements to be effective. One of the more important requirements is adequate speech intelligibility. Based on the assumption that the accepted minimum performance standard for measuring receiver sensitivity in conventional FM (12 dB SINAD) represents the minimum (or adequate) intelligibility, the values of objective performance measures for ACSB that produce the same level of intelligibility have been determined. The Articulation Score (AS) is used as the subjective voice quality measure. The experiments reported here show that the intelligibility of ACSB operating at 12 dB SINAD (and an audio-to-pilot ratio of 10 dB) is approximately the same as conventional FM operating at 12 dB SINAD.

Key words: ACSB; amplitude companding; articulation score; intelligibility; land mobile radio; SINAD; single sideband.

1. INTRODUCTION

The underlying impetus for this work is the shortage of frequency spectrum for the land mobile radio service. Because the use of less bandwidth per channel is an obvious way to reduce the crowding, there has been increased interest in recent years in "narrowband technologies." Beginning in the late 1930's, the modulation used for land mobile communications has been frequency modulation (FM) with a channel width of 25 or 30 kHz or greater. Since the transmission of intelligible voice requires a baseband from a few hundred hertz to two or four kilohertz, it seems there is indeed room for improvement regarding spectrum efficiency. On the other hand, the wide channel width used for FM contributes to the quality of the voice signal. The quest, therefore, has been and continues to be to find a modulation scheme that not only increases spectrum efficiency by using a channel width not much greater than the baseband width, but also retains the voice quality level that land mobile radio users are used to.

Single sideband (SSB) amplitude modulation with a suppressed carrier (SSBSC) has the narrowest emission bandwidth of all the analog modulation schemes. For this reason, it has been pursued—with some complex enhancements—as a solution to land mobile spectral crowding. One enhancement is the use of companding, which is achieved by first compressing the amplitude of the voice signal before modulation and transmission, and then expanding this signal after detection in the receiver to restore its original dynamic range. The salient effects of companding the baseband signal are an increase in the average transmitted power and an increase in the average received signal-to-noise ratio. Companding, along with other enhancements such as the pilot tone used on SSBSC, is called amplitude companded sideband or ACSB.

The term "compand" is the combination of the two other words: *compress* and *expand*. A compandor is the device that performs the companding functions; in the case of ACSB, the compandor comprises two distinct functional circuits—one in the transmitter and one in the receiver. It should be noted that the term "compandored" is often used and that it means the same thing as companded. The terms "compand" and "companded" are used throughout this report except in reference citations and where another author is quoted.

* Institute for Telecommunication Sciences.

1.1 Purpose and Scope

The purpose of this work is to identify the appropriate objective performance measures for ACSB and to determine the values of such measures that represent a minimum level of acceptable performance. In addition, it was hoped that sufficient test data would be obtained to develop a performance standard for testing and evaluating ACSB transceivers. There are many aspects of performance, including spectrum efficiency, propagation range, application flexibility, noise susceptibility, and voice quality. In this report, most of these are discussed, but the focus is on voice quality. The minimum level of acceptable voice quality for ACSB was chosen to be the same as the voice-quality level of accepted performance for FM. The subjective measure used to represent voice quality was chosen to be intelligibility as represented by the articulation score (AS). An objective measure of minimum performance for FM is the receiver sensitivity at 12 dB SINAD (see sec. 6.1 and app. A). Figure 1 illustrates the relationships between the objective and subjective performance measures for FM and ACSB and the unknown measures being sought in this work. At the beginning of this effort, it was not known what level of intelligibility 12 dB SINAD represented for FM.

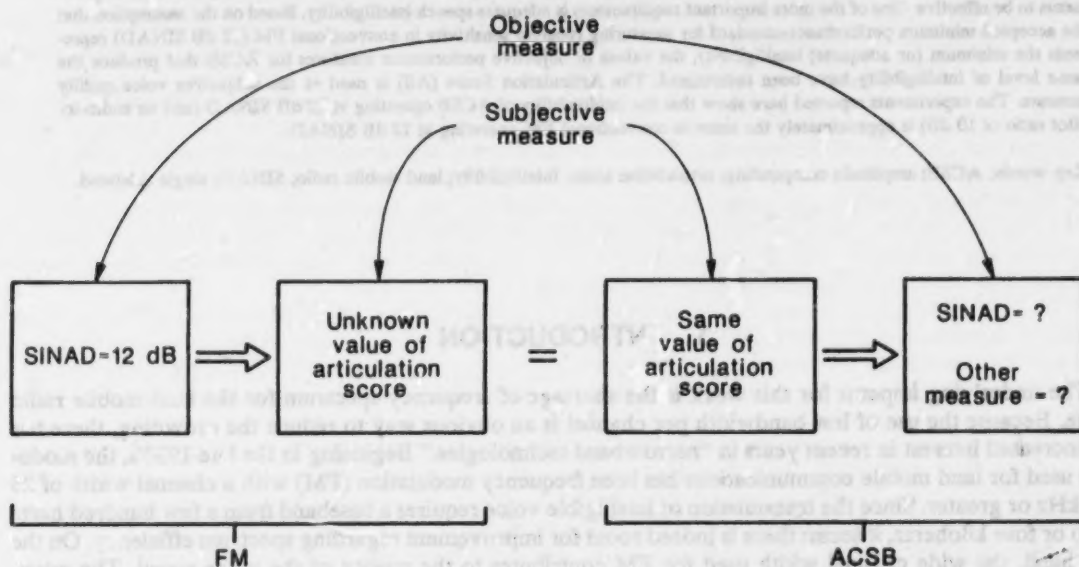


FIGURE 1. Relationships between the known and unknown measures for FM and ACSB showing the role of subjective articulation scoring.

Section 2 opens with a review of regulatory activities that allowed the introduction of narrowband technologies into land mobile communications and closes with the current state of affairs. Sections 3 and 4 present a description of ACSB and a review of relevant past work, respectively. Several installed systems observed in operation are described in section 5. Section 6 describes the measurement of SINAD and other parameters for ACSB and the measurement of the articulation score (AS) for both FM and ACSB. Overall results and conclusions are given in section 7, and a suggested performance standard for ACSB radios is being prepared as a separate document.

Appendices A and B provide more detailed information on the definition and measurement methods of SINAD and articulation score, respectively.

1.2 Brief History of Land Mobile Radio

Mobile communications refers to the situation where one or both of the terminals are located on a mobile platform that can be a boat, airplane, or land vehicle. Mobile-to-mobile links may be achieved by using a mobile-to-fixed-to-mobile link. The first practical demonstration of radio communications by Marconi in 1897 was a fixed-to-marine mobile link.

One of the first practical uses of land mobile radio communications was a dispatch service, established in 1921 for the Detroit Police Department. Another early user of land mobile radio, also for dispatch services, was the New York City Police Department. That system was established in 1932; like the earlier system,

it used the 2-MHz frequency band [1]¹. In 1938, the Federal Communications Commission (FCC) provided the rulings that defined land mobile communications services in this country. A part of these rulings was the designation of four channels in the 30/40-MHz range and a like number in the 150-MHz band. In 1956, the 450-MHz band was opened, and in 1974 the 800-MHz band was opened. At the same time, as the technology improved, the channel widths were decreased so that more users could be accommodated. The first VHF channels were 100 kHz wide. For many years they have been 25- or 30-kHz wide and now 5-kHz "narrow-band" channels are allowed in the 150-MHz band.

2. BACKGROUND AND RULEMAKING

Official action by the Federal Communications Commission (FCC) concerning ACSB commenced when it released, on September 9, 1980, a Notice of Inquiry [2] which requested comments concerning possible modification of the FCC's Rules and Regulations to allow the "introduction and use of additional technologies that can improve the efficiency of radio spectrum use." The FCC attempted to make clear that its intention was not "to mandate the use of any particular technology" but "to provide greater freedom and incentive for licensees to make selections that, while in their own best interest, also lead to greater efficiency in spectrum usage." The following were among the issues for which comments were requested:

1. What spectrum-saving technologies are suitable for voice/nonvoice use? What bandwidth and technical standards are appropriate for these technologies?
2. In what stage of development are these technologies and what time frame would be appropriate for their introduction?
3. The FCC requested comments concerning the study done on ACSB by Lusignan and others at Stanford University. Among the questions were:
 - a. Is the information in the Lusignan study useful in determining the feasibility of ACSB? What specific additions, deletions, or modifications should be made to this material?
 - b. Is ACSB suitable for land mobile use based on all of the information now available?
 - c. How does ACSB compare to FM in terms of audio quality, intelligibility, occupied and necessary bandwidths, and range (for equal power levels)?
 - d. What would be the likely cost of ACSB equipment initially? In 5 years? Would industry be interested in building ACSB equipment? Under what conditions? Should the Commission act to stimulate the design, development, production, and/or use of ACSB equipment? If so, how?
4. The FCC requested comments on whether separate spectrum assignments should be made for narrowband emissions.
5. The FCC also requested comments on whether incentives should be provided to encourage users to use spectrum-efficient technology.

Comments varied widely. Several respondents expressed the opinion that ACSB offered significant potential for improving spectrum efficiency. Others believed other existing (narrowband FM) or future (narrowband digital) technologies offered more promise and expressed doubts as to whether ACSB could operate satisfactorily in the 5-kHz spacing suggested by proponents of ACSB. Several comments suggested that ACSB should be implemented initially with developmental licenses. The reports of the study done by Lusignan on ACSB raised considerable controversy. One manufacturer of ACSB radios, Sideband Technology Incorporated (STI, now merged with Aerotron, Inc.), generally agreed with the results of the Lusignan study; the Electronics Industries Association (EIA) attacked the results. STI expressed the opinion that the price differential between narrowband and conventional equipment would disappear. Several comments suggested that separate bands should be set aside for narrowband technologies. Proponents of ACSB had indicated that ACSB could operate satisfactorily on interstitial channels between existing FM channels. STI believed that providing long-term developmental licenses, which would allow licensees to operate in the 150-MHz band on a channel with less interference, was a very strong incentive in itself. STI also suggested that minimal restrictions be placed on these licensees.

Following this Notice of Inquiry, the FCC authorized about 275 developmental licenses for ACSB. In addition, the FCC's Office of Science and Technology (OST) conducted laboratory and field tests on FM and ACSB radios [3]. Notable among its results were findings regarding co-channel and adjacent channel interference between ACSB signals and FM signals. Included were FM to FM, ACSB to FM, FM to ACSB, and ACSB to ACSB interference studies.

¹ Numbers in brackets refer to references in section 8.

On April 4, 1984, the Commission released a Notice of Proposed Rulemaking [4] to allow permanent licensing of ACSB stations. Many of the comments received to this Notice of Proposed Rulemaking were similar to those received to the Notice of Inquiry. Several respondents questioned whether ACSB could operate with sufficiently low levels of adjacent channel interference with 5-kHz channel spacings. Some manufacturers were concerned because ACSB was the only current technology that 5-kHz channels were suited to as narrowband FM emissions of 12.5 or 15 kHz would not fit into a 5-kHz channel. On March 3, 1985, the Commission took its most significant action to date involving ACSB, by adopting the Notice of Proposed Rulemaking [5] and modifying Part 90 of the FCC's Rules and Regulations to allow the permanent licensing of ACSB stations. The FCC opted for 5-kHz channels.

On March 28, 1984, the Commission released a Notice of Proposed Rulemaking [6] that proposed changes to the frequency assignment procedures in the Broadcast Remote Pickup Service, which would allow the use of ACSB. One application of remote pickup stations is the transmission of program audio from the scene of an event to the studio. Since high-fidelity sound is necessary in this application, a wide bandwidth is required. However, this service is also used for transmission of one-way or two-way operational communications to coordinate production and support technical operations of the station. The major use of frequencies in the Broadcast Remote Pickup Service is for voice-quality communications requiring only narrow bandwidths. Much of the equipment used in this latter application is the same type used by land mobile licensees. This Notice of Proposed Rulemaking's aim was to create flexibility in frequency assignments allowing the use of ACSB on 5-kHz channels resulting in greater spectrum utilization.

An alternative type acceptance procedure was proposed by the FCC in Docket 85-171 released on June 11, 1985 [7]. This docket was both a Notice of Inquiry and a Notice of Proposed Rulemaking. The proposals of this docket are in line with the approach to allow more competition in telecommunications. The only aspect of new mobile radio technologies the FCC is proposing to regulate by its alternative type acceptance is the interference these new technologies may cause. In the first paragraph of the docket the FCC states that it is striving to make interference control rules that will not be "unnecessarily restrictive in areas unrelated to interference control" and that they will "attach a low priority to mandating system operability." Changes to the FCC Rules and Regulations would be made in Parts 2, 22, 74, and 90. The change in Part 2 would include the details of the alternative type acceptance procedure. Changes would be made in Parts 22, 74, and 90 to allow the use of alternative type accepted equipment in those services. The alternative type acceptance procedure as initially proposed by the FCC is essentially a measure of the interference to FM caused by new mobile radio technologies.

3. FUNCTIONAL DESCRIPTION OF ACSB

ACSB uses amplitude companding with pre-emphasis and de-emphasis. It also uses a pilot tone as a frequency reference and for expander gain control. These signal-processing aspects and the operation of the overall system are described in this section.

3.1 Pilot Tone

The main advantages of SSBSC are its efficient use of the frequency spectrum and superior signal-to-noise characteristics compared to conventional double sideband amplitude modulation (AM). Since SSBSC uses only one of the two sidebands generated in AM, the emission bandwidth required is nominally the same as the baseband width. No carrier is transmitted in SSBSC, and the transmitter has no power output until it is modulated. This means that the power is concentrated in the sidebands, where the information is contained. The narrow bandwidth of SSBSC leads to superior signal-to-noise characteristics (for the same transmitter power) for the demodulated signal of SSBSC as compared to AM.

The absence of a carrier contributes to the superior signal-to-noise characteristics, but at the same time it is a major drawback of SSBSC. Since no carrier is received, one must be reinserted by the receiver in order to demodulate the signal, and if the reinserted carrier is not accurate, distortion of the demodulated signal will occur. The baseband will be offset in frequency by the amount of the error in the frequency of the carrier that is reinserted. An error of only 100 Hz can lead to significant distortion.

ACSB eliminates the problem by transmitting a pilot tone which provides a reference for the receiver so that it knows an ACSB signal is present and precisely where to reinsert the carrier. In some ACSB transmitters, the pilot tone is transmitted at a level equal to rated peak envelope power (PEP) for the first 250 ms after the push-to-talk switch is pressed. This 250-ms period of time is called the preamble. After the preamble, the pilot is reduced to the operating level (typically at 10 dB below rated PEP with no audio input to the transmitter) and it provides a continuing reference signal for the receiver. Transmitters that do not make use

of a preamble transmit only the pilot tone at the reduced (or operating) level. The frequency reference provided by the pilot tone makes possible the use of ACSB with fixed tuned receivers and transmitters in the mobile radio services. Radio services in the range from 50 to 450 MHz covered by the FCC Rules and Regulations Part 90 (police, fire, business, taxi, etc.), have a transmitter frequency tolerance of 0.0005 percent (5 ppm) for all fixed stations and for mobile stations that have a power of 2 W or more. For 5-kHz channels, the revised Part 90 states that the frequency tolerance shall be 0.0002 percent (2 ppm). This corresponds to an error of 300 Hz at 150 MHz. An error of 300 Hz in the insertion of the carrier would yield a demodulated signal that is probably unintelligible to most people. However, the ACSB pilot tone provides a reference for the receiver so that it is not necessary to have an inordinately small frequency tolerance for fixed-tuned ACSB transmitters and receivers.

The pilot tone may be put in one of three locations relative to the speech pass band:

- tone-above-band
- tone-in-band
- pilot carrier.

Tone-above-band inserts the pilot tone above the speech information while tone-in-band inserts the pilot tone in the speech passband. Pilot carrier inserts the pilot tone at the carrier frequency. A pilot carrier allows for a simpler receiver; however, it has the disadvantage of causing more problems with adjacent-channel interference than either of the other two formats. The reason for this is that the carrier would be close to the higher-frequency voice components of the lower adjacent channel. The higher frequency voice components generally have less energy and are therefore more susceptible to interference. The ACSB equipment currently manufactured in the United States uses a tone-above-band pilot at 3100 Hz to avoid these problems. Figure 2a illustrates an idealized frequency spectrum of an ACSB transmitter in the currently accepted format. Figure 2b illustrates the frequency spectrum of the same transmitter, but with a 1-kHz audio input.

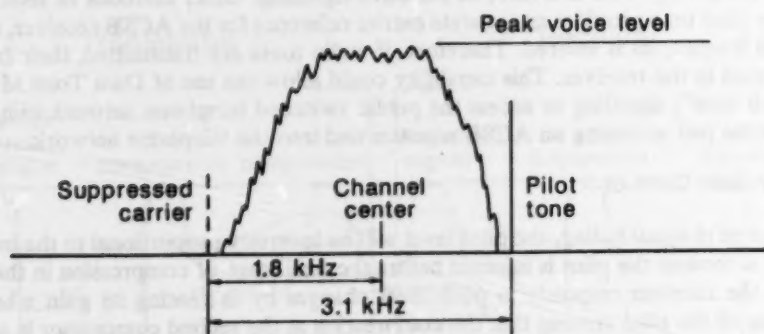


FIGURE 2a. Idealized emission spectrum of an ACSB signal illustrating the envelope of voice modulation. The nominal range of the voice band is from 300 to 3000 Hz.

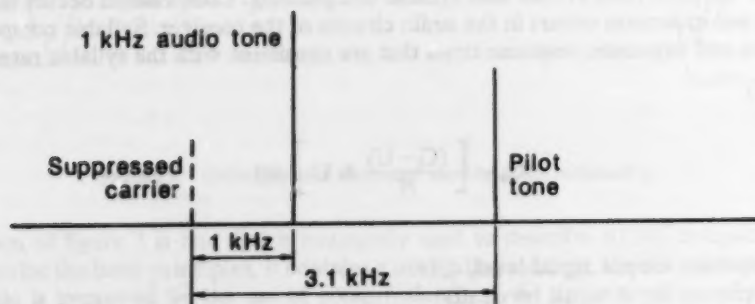


FIGURE 2b. Idealized emission spectrum of an ACSB signal modulated with a 1-kHz tone.

Besides providing an accurate frequency reference for the receiver, the pilot tone also may provide automatic gain control at rf and/or IF, squelch, selective signaling, and expander gain control (also called "pilot pumping" or feed forward gain control).

3.1.1 Automatic Gain Control (AGC)

Since no carrier exists to provide a reference for AGC circuits, the pilot tone provides such a reference. Following some filtering and therefore some averaging, the pilot level is used to control the gain of one or more rf and IF amplifiers. Since the gain of the first expander is controlled by the pilot level, it also provides AGC action.

3.1.2 Squelch

The presence of the pilot tone allows for positive squelch action in the receiver. In a simple squelch system the audio of the receiver would be shut off until the presence of a pilot tone was detected. ACSB radios can use the Continuous Tone Coded Squelch System (CTCSS) that is commonly provided with FM radios. The pilot tone can be frequency modulated at a low level (with a modulation index of 1.6 to 1.8) with frequencies in the range of 70 to 200 Hz. Individual frequencies in this range may be assigned to different users so that the squelch on their receivers will be disabled only if the pilot tone is being modulated with the frequency assigned to the particular user. In this way, multiple users are able to share the same frequency without being annoyed by hearing the transmissions of other users.

3.1.3 Selective Signaling

In the squelch circuits mentioned above, the squelch is disabled only when the pilot tone is modulated with a particular frequency. This is a form of selective signaling. Other methods of selective signaling are possible. Since the pilot tone provides an accurate carrier reference for the ACSB receiver, accurate recovery of the transmitted frequencies is assured. Therefore, if audio tones are transmitted, their frequencies will be faithfully reproduced in the receiver. This capability could allow the use of Dual Tone Multiple Frequency (DTMF or "touch tone") signaling to access the public switched telephone network using an ACSB radio equipped with a tone pad accessing an ACSB repeater tied into the telephone network.

3.1.4 Expander Gain Control

In the absence of rf signal fading, the pilot level will be inversely proportional to the instantaneous voice signal level. This is because the pilot is injected before the final stage of compression in the transmitter. The first expander in the receiver responds to pilot level changes by increasing its gain when the pilot level decreases. This use of the pilot ensures that the compression at the second compressor is accurately undone by the expansion of the first expander (see sec. 3.2). If an rf fade occurs, both the pilot and the voice signal will decrease and the first expander's gain will increase, thereby counteracting some of the fading effects.

3.2 Companding

In addition to the pilot tone, ACSB uses syllabic companding. Compression occurs in the audio circuits of the transmitter and expansion occurs in the audio circuits of the receiver. Syllabic companding implies the use of compression and expansion response times that are consistent with the syllabic rates in speech. In the compressor

$$C_o = \left[\frac{(C_i - U)}{N} + U \right] \text{ dB} \quad (1)$$

where C_o = compressor output signal level, dB

C_i = compressor input signal level, dB

U = unaffected level, dB. This is the input level at which the output level equals the input level. Put another way, this is the input level at which the compressor has 0 dB gain. This level is labeled 0 dB in figure 3.

N = compression ratio.

In the expander the reverse process takes place and the compressed signal is restored to its original dynamic range. In the expander,

$$E_o = [(E_i - U) \times N + U] \text{ dB} \quad (2)$$

where E_o = expander output signal level, dB

E_i = expander input signal level, dB

U = unaffected level, dB

N = expansion ratio.

From the above expressions it can be seen that the dynamic range (in dB) of the signal is reduced by a factor equal to the compression ratio. The currently accepted ACSB system uses a pair of 2-to-1 compressors in the transmitter and a pair of 1-to-2 expanders in the receiver. Figures 3 and 4 illustrate this companding with the introduction of noise in the transmission path. The dynamic range of the signal (consider it to be a syllable) is 30 dB at the input to the first compressor. The dynamic range of the signal leaving the first compressor is 15 dB. Next, the signal enters the second compressor. The signal leaving the second compressor has a dynamic range of only 7.5 dB, one-fourth of that entering the first compressor. Note that the effect of reducing the dynamic range of the signal is to significantly increase the median level.

For the original signal, the median level is 15 dB below the maximum level. After undergoing two stages of compression, the median level is only 3.75 dB below the maximum level—an increase in the median level of 11.25 dB. Raising the median level of the audio signal with respect to the noise in the transmission path makes the compressed signal less susceptible to that noise. During expansion the original amplitude of the signal is restored.

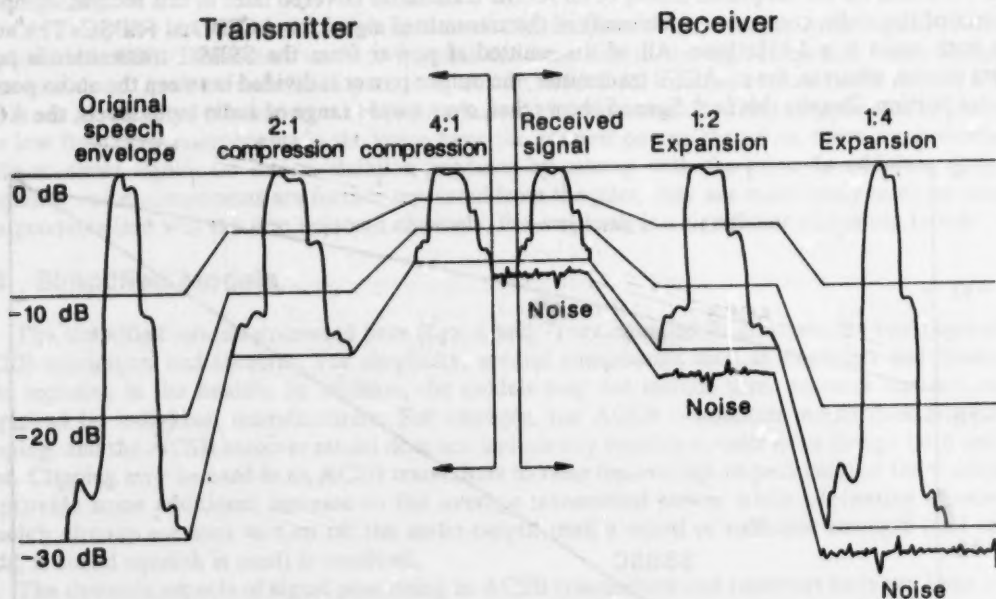


FIGURE 3. Commonly used illustration showing ACSB companding.

The illustration of figure 3 is one that is commonly used to describe ACSB companding. Although it serves well to describe the basic principles, it contains a misleading subtlety. As discussed above, the received signal-to-noise ratio is improved by the use of compression, and the dynamic range of the audio signal is restored by expansion. It is important to note that expansion does not improve the signal-to-noise ratio because the instantaneous expander gain must be applied to both the signal and the noise at the same time, as shown in figure 4.

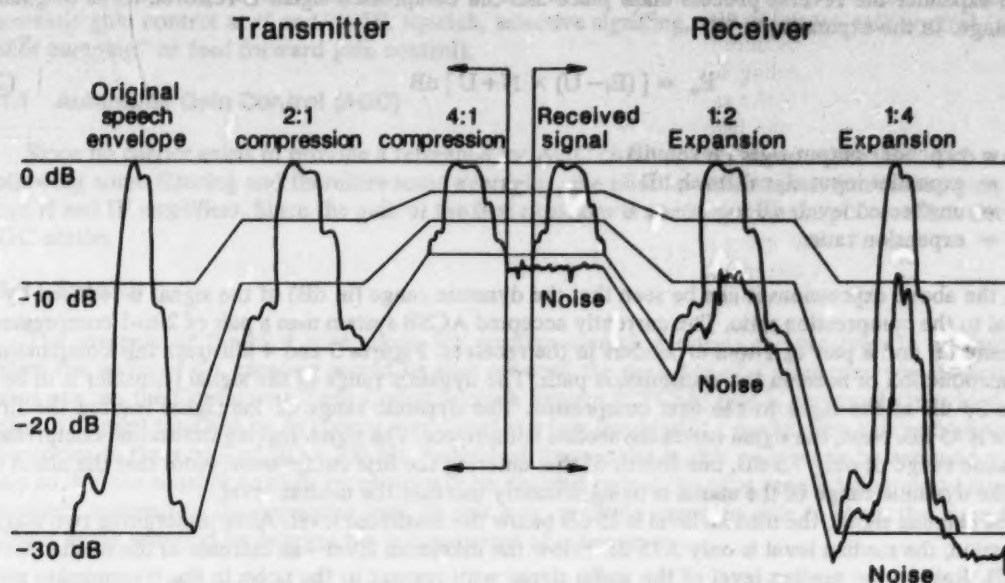


FIGURE 4. Actual effect of ACSB expansion (refer to fig. 3).

Figure 5, based on the simplified model of an ACSB transmitter covered later in this section, compares the rf power of the audio component (sideband) of the transmitted signal for ACSB and SSBSC. The audio input for both cases is a 1-kHz tone. All of the emitted rf power from the SSBSC transmitter is audio component power, whereas, for an ACSB transmitter, the output power is divided between the audio portion and the pilot portion. Despite this fact, figure 5 shows that, over a wide range of audio input levels, the ACSB

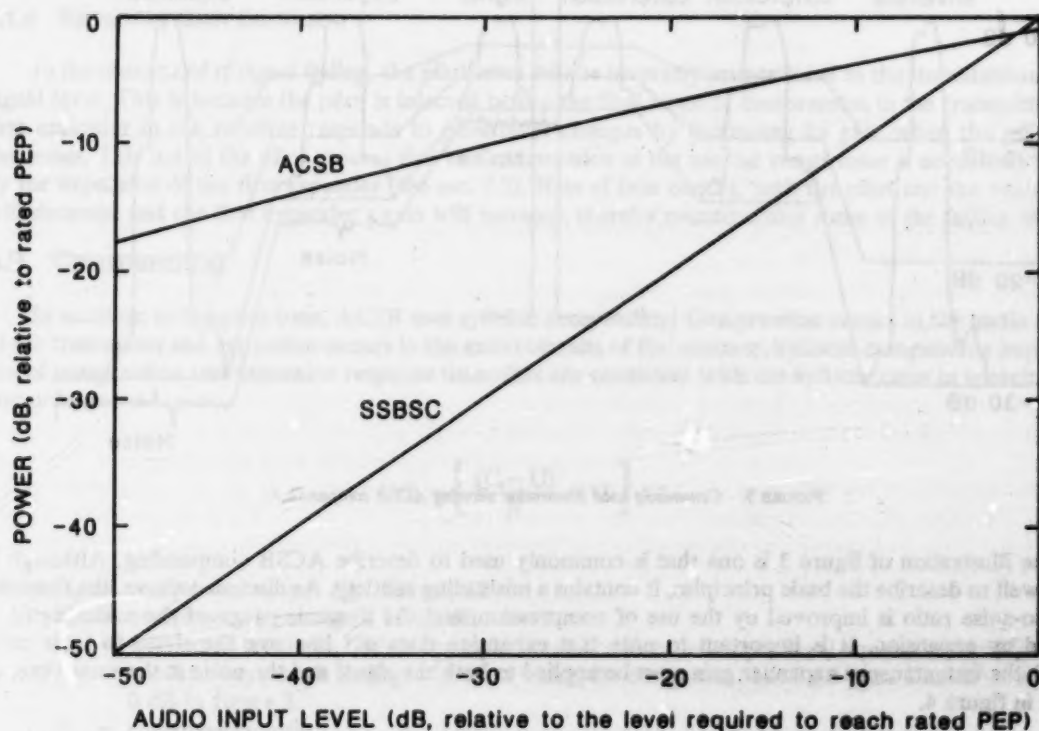


FIGURE 5. The rf power contained in the information sidebands of ACSB and SSBSC vs. audio input level.

transmitter produces considerably more rf power in the audio component than does SSBSC. It can be seen from the above discussion and figure 5 that compression reduces the effects of noise, especially for lower level signals. Expansion greatly reduces the noise during pauses in speech. All of these effects taken together lead to what is called the companding advantage. When a system using companding is operated on a channel with noise, the subjective effect is the same as a system not using companding on a channel with less noise. This advantage is dependent on the listener, the volume and nature of the speech source, and the amount and nature of the noise. Experiments done at Rensselaer Polytechnic Institute [8] revealed that the companding advantage may be considered to be a random variable. The investigation showed there was a 90 percent probability that 2-to-1 companding provided at least a 16-dB companding advantage.

It has been shown that using companding with FM can significantly increase the capacity of satellite transponders. An even greater increase is possible using ACSB. Bell Laboratories investigations [9] show that up to 7800 voice circuits per transponder on TELSTAR 3 are possible by using ACSB—four or five times that which is possible using FM. A significant factor in this increased capacity is ACSB's ability to reduce the effect of link noise. An even greater increase in capacity would be possible if the circuits were used strictly for voice, because the subjective effects of companding are greatest for voice. Since AT&T desired that all these voice circuits be capable of handling data rates of up to 4800 bps on unconditioned lines, the increase in capacity is limited to four or five times that possible using FM.

3.3 Pre-emphasis and De-emphasis

ACSB transceivers also make use of pre-emphasis (in the transmitter) and de-emphasis (in the receiver). The higher frequency components of voice are lower in amplitude than the low frequency components. Even though the higher frequency components are lower in power, they are very important to intelligibility. The process of pre-emphasis and de-emphasis is used routinely in FM transceivers. The primary reason that it benefits FM is that the noise power spectrum of a discriminator (FM detector) increases with the square of the frequency.

Obviously the ACSB receiver does not use a discriminator to recover its audio signal. Although pre-emphasis in ACSB transmitters tends to increase the immunity of the high frequency audio components to noise introduced in the transmission path, it is mainly needed for another reason. Since ACSB is a form of amplitude modulation, it is important to reduce intermodulation resulting from components in its own signal. The low frequency components in the voice have the greatest power; therefore, these components are most likely to cause significant intermodulation products by mixing with the pilot. In addition, since the low frequency voice components are further separated from the pilot, they are more likely to cause intermodulation products that will fall into adjacent channels. Pre-emphasis is a significant mitigating factor.

3.4 Simplified Models

The simplified models presented here (figs. 6 and 7) are intended to illustrate the basic operation of an ACSB transmitter and receiver. For simplicity, several components such as amplifiers and filters have not been included in the models. In addition, the models may not include some features that are, or may be, employed by individual manufacturers. For example, the ACSB transmitter model does not include any clipping, and the ACSB receiver model does not include any squelch circuits even though both are currently used. Clipping may be used in an ACSB transmitter to raise the average-to-peak ratio of the voice signal and to provide some additional increase to the average transmitted power while preventing overmodulation. Squelch circuits are used to turn off the audio output until a signal of sufficient strength (and appropriate code, if coded squelch is used) is received.

The dynamic aspects of signal processing in ACSB transmitters and receivers have not been included in the simplified models or in any of the descriptions presented in this section. This complicated aspect of ACSB is very important to its performance. The compressors have time constants, as do the expanders. For example, one might expect the first expander to have a very short time constant so its gain can accurately follow the pilot level. The time constant of the second expander may be such that it responds to each syllable of speech. The time constants for rf and IF AGC might be much longer so that their gains are controlled by rf signal fades rather than by each syllable of speech. Clipping circuits used in the transmitter to prevent overmodulation and distortion may be soft clippers, that is, a dynamic signal is allowed to exceed the level at which a static signal is held. These and other time constants contribute to the nature and performance of ACSB. For this discussion, the signal levels are static or quasi-static, so any effects due to signal dynamics are excluded.

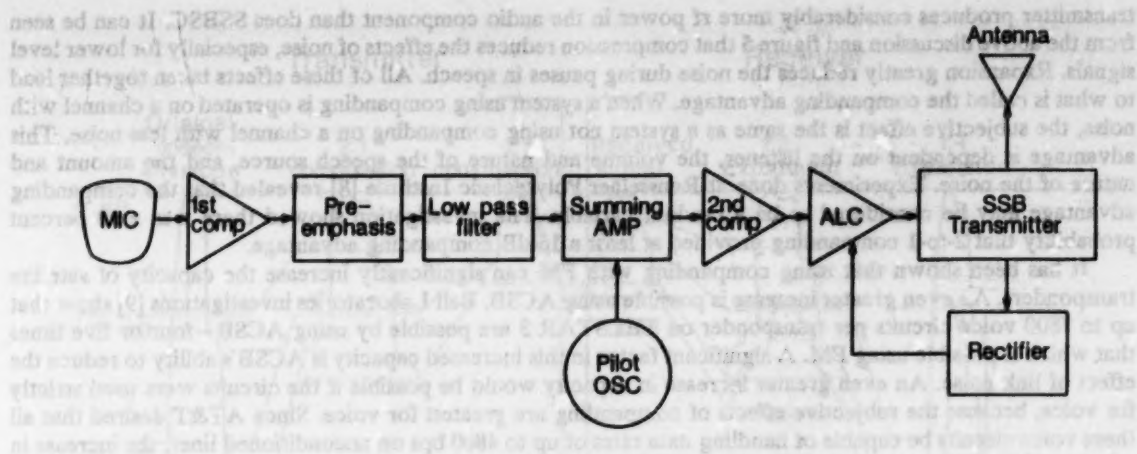


FIGURE 6. Simplified ACSB transmitter block diagram.

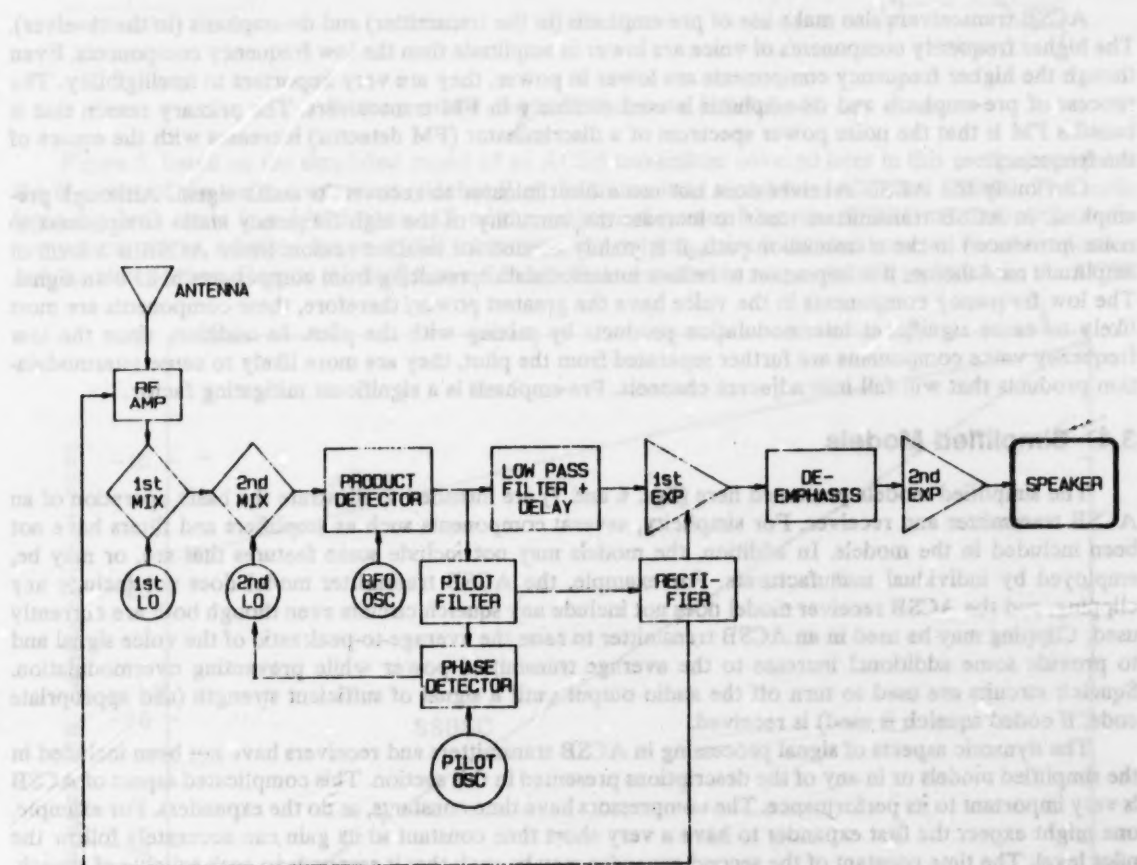


FIGURE 7. Simplified ACSB receiver block diagram.

3.4.1 Transmitter (refer to fig. 6)

1. Audio signal passes from the microphone into the first compressor where it undergoes 2-to-1 compression.
2. Audio signal enters the pre-emphasis circuit where it undergoes 6-dB/octave pre-emphasis.
3. Low pass filter removes any audio signals above about 2500 Hz to provide for the pilot.
4. Audio signal and the pilot (3100 Hz) are combined in a summing amplifier.
5. Audio signal and pilot enter the second compressor, where the composite of audio and pilot signals undergoes 2-to-1 compression. The reason for compressing the audio signal and the pilot together is that the pilot level after compression will contain information about the amount of compression. When the audio level is low, the compressor gain is high, so the pilot level is high. When the audio level is high, the compressor gain is low, so the pilot level is low. The result is that the instantaneous pilot level is proportional to the compressor gain. This effect is called "pilot pumping" and is used to drive the first expander in the receiver.
6. Automatic level control (ALC) circuit receives feedback from the rf output so that it can restrict the audio peaks to prevent the rf output from being significantly greater than the rated PEP. This prevents the rf amplifier from being driven into a nonlinear region.
7. Audio and pilot then enter rf circuitry where they modulate the rf signal to be transmitted.

3.4.2 Receiver (refer to fig. 7)

1. The rf signal and first local oscillator (LO) signals are mixed in the first mixer to form the first IF.
2. First IF signal and second local oscillator signals are mixed in the second mixer to form the second IF.
3. Second IF signal and beat frequency oscillator (BFO) are mixed in the product detector to demodulate the signal. The BFO mixing is the reinsertion of the missing carrier.
4. Demodulated signal is split into two paths following the product detector.
5. Part of the demodulated signal passes through the pilot filter where the pilot signal (3100 Hz) is filtered out of the demodulated signal.
6. Part of the pilot signal is used to help perform an automatic frequency control (AFC) function. This part of the pilot goes to a phase detector where its frequency is compared with that of the pilot oscillator in the receiver. The frequency difference between these two signals causes a voltage to be generated that is used to tune the second oscillator so that the received pilot and the locally generated pilot are identical in frequency.
7. Another part of the pilot signal is used to drive the first expander and to help perform an automatic gain control (AGC) function. This portion of the pilot goes through a rectifier. The first expander restores the pilot to a fixed level, so the gain of the first expander is inversely proportional to the pilot level. In this way the first expander not only undoes the compression done by the second compressor, but it also works as an AGC. An increase or decrease in the pilot level, whether due to compression in the second compressor of the transmitter or due to a change in the level of the rf signal received, will cause the gain of the first expander to increase or decrease to compensate for the change in pilot level. This technique is called feed forward gain control, pilot pumping, or expander gain control.
8. Rectified pilot can be used to control the gain of the rf and/or IF amplifiers to provide more AGC action.

9. The demodulated signal also passes through a low pass filter and delay circuit. The purpose of the low pass filter is to allow only the audio signal to pass. The narrow filter in the pilot path decreases the noise on the pilot and increases the sensitivity of the receiver; however, a narrow pilot filter also introduces more delay than the low pass filter the audio must pass through. In order for the pilot that controls the gain of the first expander and its associated audio to be kept in synchronization, some delay of the audio must be introduced.
10. Audio signal enters the first expander.
11. Audio signal leaves the first expander and enters the de-emphasis circuits where it undergoes 6 dB/octave de-emphasis undoing the 6-dB/octave pre-emphasis done in the transmitter.
12. Audio signal then undergoes 1-to-2 expansion in the second expander.
13. Audio signal is amplified to drive the receiver's speaker.

3.4.3 Simulated Signal Levels

While the models just described may differ somewhat from transceivers produced by individual manufacturers, they provide a useful basis for understanding the principles of ACSB. The results that follow are based on this model of an ACSB transmitter. It is assumed that the pilot rf power produced (following any preamble, if there is one) when the transmitter is keyed with no audio input is 10 dB below rated PEP.

Figure 8 illustrates how the compression of the audio signal in the transmitter varies with input level. The input signal is a 1-kHz tone. The 0-dB audio input level is the audio input that will produce an rf output power equal to rated PEP. All audio input levels are referenced to this level. Note that at the lowest audio input level shown in figure 8 the compression ratio is about 2.4-to-1 and even at an audio input level producing rated PEP the compression ratio is about 3.63-to-1. The reason that compression of the input audio signal is less than 4-to-1 is that the pilot tone is combined with the audio prior to the second compressor.

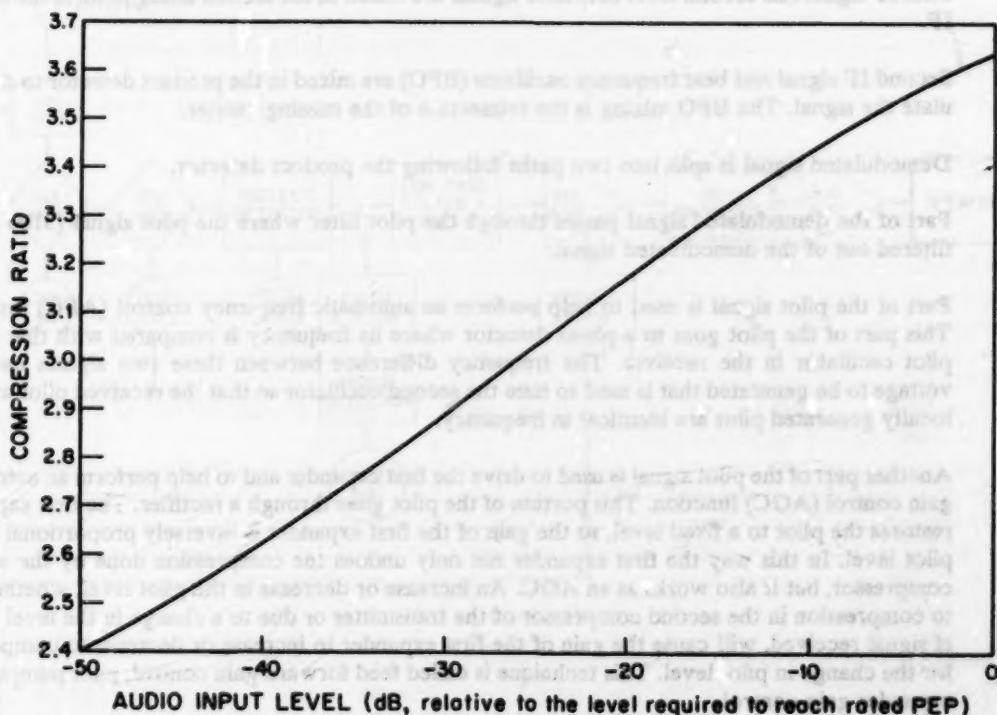


FIGURE 8. Compression ratio vs. audio input level.

Before the audio signal enters the second compressor, a fixed-level pilot tone at -20 dB, relative to the level of input into the second compressor required to produce rated PEP, is added to it. Since the input to the second compressor is a combination of the pilot and the audio signal from the first compressor, its gain is reduced from what it would be if its input were only the audio signal. Figure 9 shows how the existence of the pilot affects the gain of the second compressor. The gain values in figure 9 have been normalized so that the lowest gain for the second compressor (when the pilot has been added to input audio) is unity. As might be expected, adding the pilot has the greatest effect on the gain of the second compressor for low audio signal levels. Conversely, the addition of the pilot to the audio signal has the least effect on the gain of the second compressor for high audio signal levels. As the audio input level increases the pilot has less influence on the second compressor gain. Therefore, as illustrated by figure 8, the compression ratio increases as the audio input level increases.

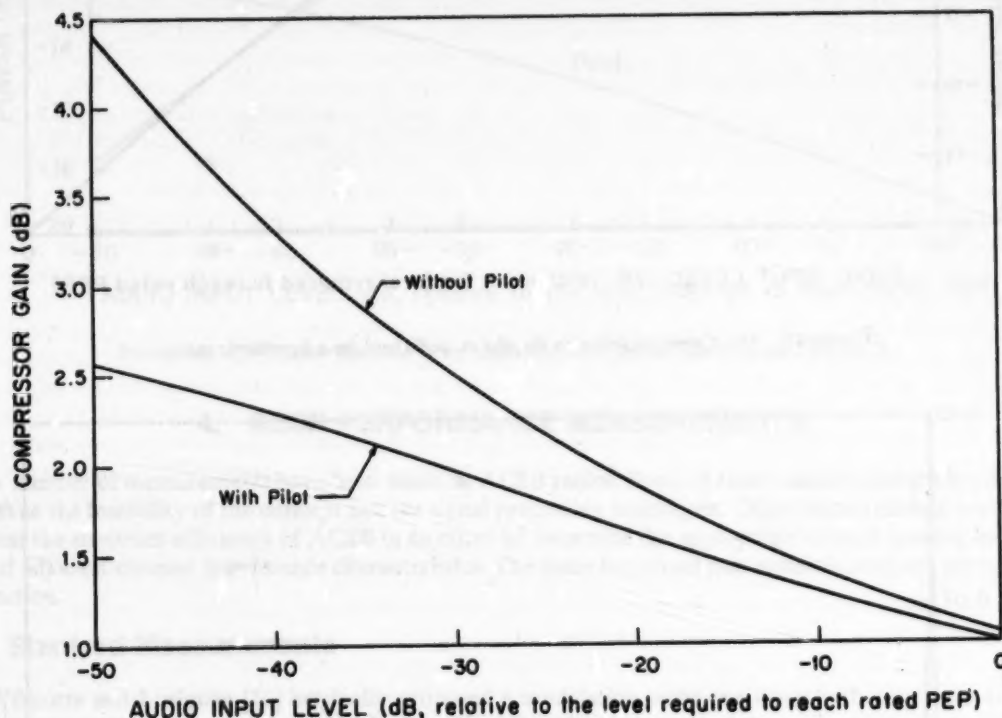


FIGURE 9. Normalized gain of the second compressor vs. audio input level.

Figure 10 shows how the pilot power varies with the audio input level. Pilot power is at its maximum when there is no audio input. Figure 11 is the same as figure 10, except with a linear scale for pilot power instead of a logarithmic scale. Figure 12 shows how both the audio power and pilot power vary with the audio input level.

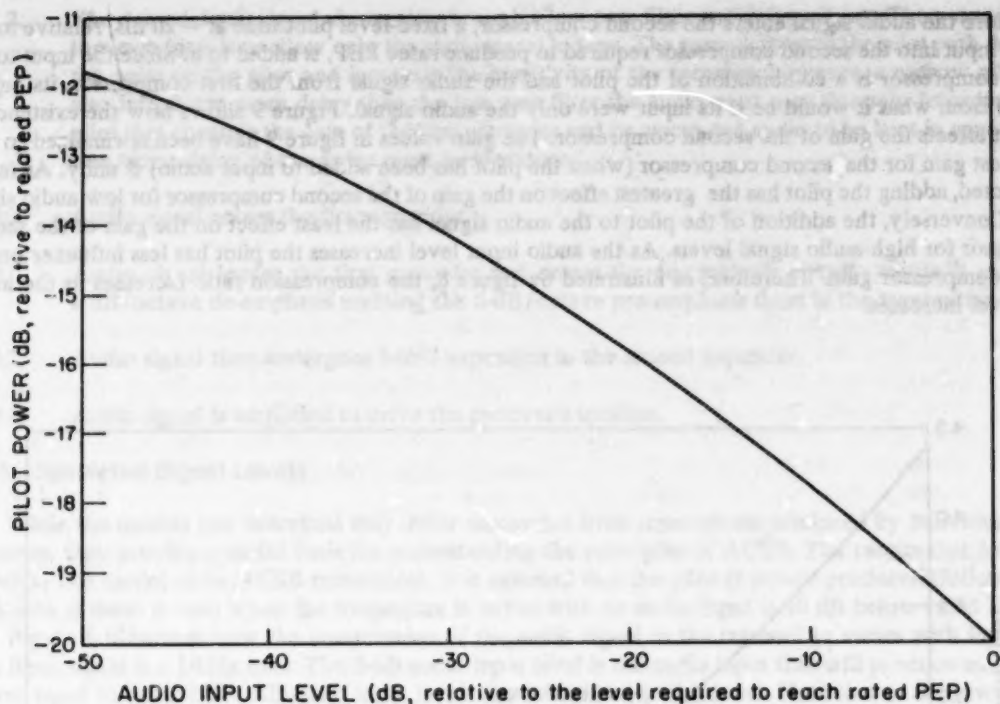


FIGURE 10. The rf power contained in the pilot vs. audio level (on a logarithmic scale).

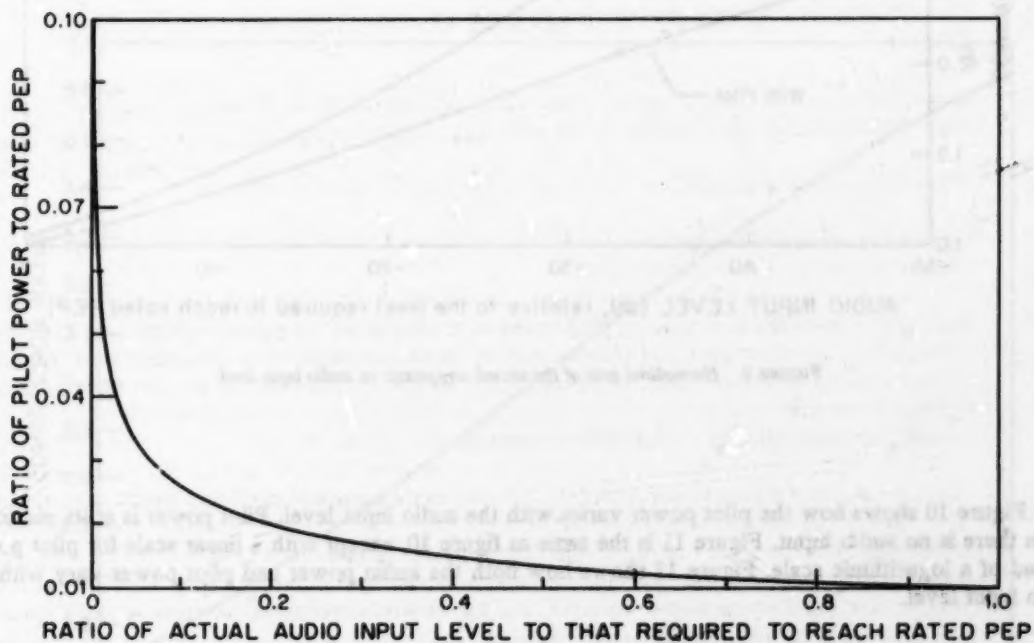


FIGURE 11. The rf power contained in the pilot vs. audio input level (on a linear scale).

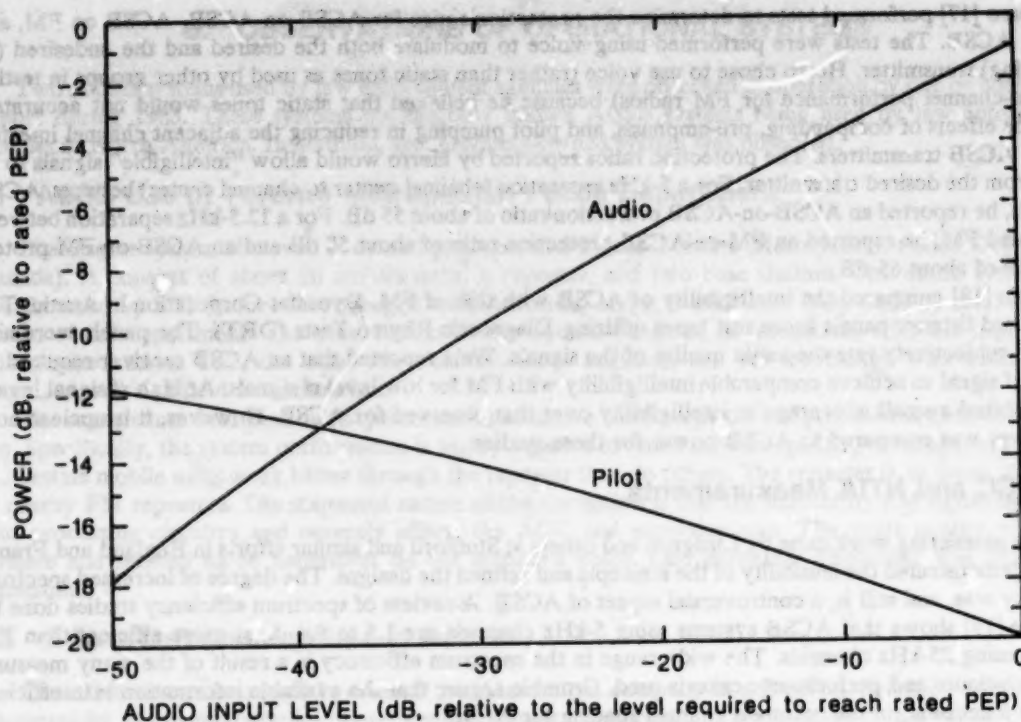


FIGURE 12. The rf power contained in the audio component and the pilot vs. audio input level.

4. ACSB PERFORMANCE MEASUREMENTS

A number of measurements have been made on ACSB radios. Some of these measurements were done to determine the feasibility of the concept and the signal processing techniques. Other measurements were made to assess the spectrum efficiency of ACSB in an effort to determine the appropriate channel spacing based on co- and adjacent-channel interference characteristics. The more important past measurements are reviewed in this section.

4.1 Stanford Measurements

Wilmette and Lusignan [10] originally proposed a modulation technique using both amplitude and frequency companding. Frequency companding performed a "folding over" of the audio spectrum to decrease the bandwidth required to transmit voice information. A technique using both amplitude and frequency companding would have been more spectrum efficient, but frequency companding was dropped because it created excessive distortion.

An early version of ACSB had a pilot tone transmitted at a power of 7 dB below rated PEP with no audio input. Current ACSB transmitters use a pilot 10 dB below rated PEP (following the preamble if there is one) with no audio input. The early version of ACSB also had a narrower baseband width, with the pilot at 2850 Hz.

Lusignan [11-15] and others at Stanford studied these earlier versions of ACSB.

Nocedal [16] performed laboratory tests to measure the advantages of companding. He concluded that SSBSC required about a 22 dB peak power-to-noise ratio for an intelligible signal whereas ACSB required a 6 dB peak power-to-noise ratio for an equivalent signal.

Nocedal developed computer models to predict the intermodulation performance of ACSB transmitters using different types of audio processing—companding, clipping, and pre-emphasis. Reducing intermodulation products allows for a closer geographic spacing of adjacent channel users and therefore has a significant impact on spectrum efficiency. These studies showed that intermodulation products falling into adjacent bands could be reduced by using pre-emphasis and pilot pumping. Tests on ACSB transmitters indicated that pre-emphasis and pilot pumping reduced adjacent channel intermodulation products by 12 dB.

Herro [17] performed tests to determine the protection ratios for ACSB on ACSB, ACSB on FM, and FM on ACSB. The tests were performed using voice to modulate both the desired and the undesired (or interfering) transmitter. Herro chose to use voice (rather than static tones as used by other groups in testing adjacent-channel performance for FM radios) because he believed that static tones would not accurately show the effects of companding, pre-emphasis, and pilot pumping in reducing the adjacent channel interference of ACSB transmitters. The protection ratios reported by Herro would allow "intelligible" signals to be heard from the desired transmitter. For a 5-kHz separation (channel center to channel center) between ACSB channels, he reported an ACSB-on-ACSB protection ratio of about 55 dB. For a 12.5-kHz separation between ACSB and FM, he reported an FM-on-ACSB protection ratio of about 58 dB and an ACSB-on-FM protection ratio of about 65 dB.

Wala [18] compared the intelligibility of ACSB with that of FM. Dynastat Corporation in Austin, TX, had trained listener panels score test tapes utilizing Diagnostic Rhyme Tests (DRT). The panels were also asked to subjectively rate the audio quality of the signals. Wala reported that an ACSB receiver required 10 dB less rf signal to achieve comparable intelligibility with FM for low level rf signals. At high rf signal levels, FM exhibited a small advantage in intelligibility over that observed for ACSB. However, it is unclear how FM power was compared to ACSB power for these studies.

4.2 FCC and NTIA Measurements

The pioneering work done by Lusignan and others at Stanford and similar efforts in England and France clearly demonstrated the feasibility of the concepts and refined the designs. The degree of increased spectrum efficiency was, and still is, a controversial aspect of ACSB. A review of spectrum efficiency studies done by Crombie [19] shows that ACSB systems using 5-kHz channels are 1.5 to 8.4 times more efficient than FM systems using 25-kHz channels. The wide range in the spectrum efficiency is a result of the many measurement techniques and performance criteria used. Crombie argues that the available information is insufficient to make a decision on the optimum channel spacing for ACSB.

Following this pioneering work, the U.S. Government was offered an opportunity by Sideband Technology, Inc., to test the production systems they manufacture. Based on that offer, a joint measurement program by the FCC and the National Telecommunications and Information Administration (NTIA), Spectrum Engineering and Analysis Division was planned [20]. The joint test program was designed to verify the technical characteristics of the radios, perform subjective laboratory tests in a simulated land mobile environment, and perform subjective field tests to demonstrate the operation in an actual land mobile environment. The joint test program was never carried out; instead, the two agencies conducted individual test programs.

The FCC's Office of Science and Technology [3] performed a laboratory and field testing program. The goals of that work were to determine the protection ratios for cochannel, adjacent channel, and drop-in interference situations and to assess performance under various conditions. The resultant information helps to answer the questions of how ACSB affects the existing FM environment, what the reuse potential is for ACSB channels, and what ACSB channel spacing is required.

The measurement program pursued by NTIA's Spectrum Engineering and Analysis Division had goals similar to the FCC's, and it included a comparison with another narrowband system—12.5 kHz FM. The NTIA work [21] specifically addressed three areas for conventional FM, ACSB, and narrowband FM. These areas were operation and use, spectrum efficiency, and implementation. As part of the NTIA program, the Electromagnetic Compatibility Analysis Center performed adjacent signal interference tests on ACSB radios [22]. Another part of the NTIA program was an analytical and empirical examination of the effects of bandwidth and interference rejection on spectrum efficiency [23].

The relevant conclusions of the NTIA work are:

1. The use of average power for FM and peak envelope power (PEP) for ACSB is appropriate and fair for specification and comparison.
2. The subjective voice quality for FM and ACSB was similar in a high signal environment. However, in a low signal environment, the voice quality of ACSB is "not as good as" conventional FM. The reasons are that "the ACSB voice signal is subject to distortion probably due to the companding, pilot tone circuitry, and/or reduced audio bandpass."

5. OBSERVATIONS OF OPERATIONAL SYSTEMS

Two ACSB systems used by law enforcement agencies were observed in operation by one or both of the authors during 1985. These systems are the only known ACSB systems being used by law enforcement agencies in North America. In addition, one privately-owned system was observed.

5.1 ACSB Use at Toronto Metropolitan Police Department

The largest system observed is owned and operated by the Toronto Metropolitan Police Department (Canada). It consists of about 20 mobile units, a repeater, and two base stations. The entire system was manufactured by STI (before they merged with Aerotron) and it is used on a daily basis by the Summons Bureau of the department. The observations of this system consisted of discussions with the head of the Telecommunications Unit, the chief telecommunications engineer, and the technicians who maintain the system, as well as a field demonstration.

The experiences described by the personnel listed above were verified during the 6-hour field demonstration. Specifically, the system performance is highly variable in terms of voice quality, coverage, and reliability. Certain mobile units work better through the repeater than do others. The repeater is, at times, disrupted by nearby FM repeaters. The suspected nature of the disruption is that the interfering FM signal enters the pilot processing circuitry and severely affects the AGC and expander gain. The voice quality was quite variable and seemed to be dependent on the speaker's voice. One particular dispatcher was difficult to understand.

5.2 A Law Enforcement Use in Denver

The other system is owned by a small communications company in a Denver, Colorado suburb and is often used by a community police department for covert operations. The observation of this system consisted of discussions with the proprietor and a 2-hour field demonstration. The base station and several mobile units that comprise this system were manufactured by Stephens Engineering Associates (SEA). This system appears to be operating very well. The coverage is as good or better than a collocated FM system. The voice quality is described by the owner and the police officers who have used it as being "somewhat mechanical." The owner reported that the users have never complained about the intelligibility of the voice, and that they consider this aspect to be equivalent to conventional FM. The short field demonstration verified the descriptions of voice quality.

5.3 A Private System in Culver City

The only private system observed is owned by Riordan Plumbing in Culver City, CA. The main reason that ACSB was selected over FM was because the owner of the company wanted to own the radio system. The only FM channels available were on a trunked system and he did not want to pay a fee for each transmission. The system is comprised of five mobile units and a base station manufactured by SEA and purchased from Mobile Communications, Inc. Installation was completed in August 1985. The observations consisted of discussions with the owner of Riordan Plumbing and several employees who use the system regularly and a 1-hour field demonstration and discussions with Mobile Communications regarding system maintenance and problems.

The owner and users reported problems with the mobile units attributable to the vibration environment. Most of the failures were broken wires on coils, etc. However, some of the problems, probably from maladjustment of certain filters and oscillators, caused significant audio distortion. Following repairs and tune-up, there were no complaints about voice quality, but a significant difference was observed between the male and the female dispatcher—more "distortion" was noted in the female voice. This was verified in the short field test, even though it was not considered a problem. Conversations with the technician at Mobile Communications verified that there were two classes of problems. One of these, probably caused by vibration, was broken wires that were usually in the oscillator sections. The other was the critical alignment of modulation of the pilot for the continuous tone coded squelch. If the level is set too low (below manufacturers specifications), the tone coded squelch may not function. If set too high, serious distortion of the demodulated audio can result from incorrect recovery of the pilot frequency by the receiver.

6. MEASUREMENTS AND ANALYSES

The purpose of setting minimum performance specifications for equipment is to provide the information needed so that the user can easily determine if that equipment meets certain requirements. One of those requirements—receiver sensitivity—is based on received signal quality usually expressed by some form of signal-to-noise ratio. As explained in section 1.1 and illustrated in figure 1, such a measure of performance is easy to measure and easy to document in a system specification. However, that objective measure must be calibrated with a different, user-oriented performance measure that better describes what ultimately matters—the intelligibility of the human voice reproduced at the receiver. Voice intelligibility measures are part of a larger category called subjective performance measures, which are too expensive to be used for routine specification testing.

Using ACSB radios borrowed from the only two companies in the United States that manufacture them, both the subjective and objective measurements described above were performed. In addition, a number of other measurements were made in support of the development of performance standards for ACSB equipment.

6.1 Objective Measurements

SINAD—the objective measure to be used in these tests—is the acronym for

$$\text{SINAD} = 10 \log \left[\frac{s+i+n+d}{i+n+d} \right] \text{ dB} \quad (3)$$

where s = signal power, W
 i = interference power, W
 n = noise power, W
 d = distortion power, W.

In most cases, the signal is a 1-kHz sine wave that is used as input to the transmitter. For FM, the tone is injected at a level producing 3-kHz deviation. At the output of the receiver, a very narrow notch filter centered at 1 kHz is used to obtain the denominator of the SINAD. SINAD may be defined in terms of a signal of different frequency and/or different input amplitudes if necessary. In any case, if the methods of making the measurements are specified, SINAD represents a measurement that is repeatable. SINAD is the simplest form of baseband signal-to-noise ratio that includes interference, noise, and distortion. Appendix A gives a more detailed description of SINAD and methods used to measure it. There are more complicated signal-to-noise measures and these are discussed below, but they are generally too complex or expensive for routine measurement.

The Articulation Index (AI) [24] is a measure using the average of weighted signal-to-noise ratios over 20 frequency bands covering that part of the audio spectrum important for intelligibility. The AI is usually easy to correlate with Articulation Score (AS) for a variety of signal processing conditions; however, it is difficult to measure and calculate without special equipment. The measurement of AI directly using a single test instrument has been approximated in two independent developments. The first of these systems was developed in about 1960, and is used by the U.S. Army Electronic Proving Ground at Ft. Huachuca, Arizona. It is called "The Voice Interference Analysis System" (VIAS).

In the VIAS method of computation, the speech frequencies between 200 and 6100 Hz are divided into 14 equally contributing bands. The signal-to-noise ratio in decibels in each band is scaled so that unity corresponds to a signal-to-noise ratio of +18 dB, and zero corresponds to a signal-to-noise ratio of -12 dB. Ratios above or below these values are rated as one or zero, respectively. The individual band contributions are summed and divided by the number of channels used, and the result is the approximation to AI.

The second system developed for automatically measuring the AI is the Speech Communication Index (SCI). This development was sponsored by the U.S. Air Force, Electronic Systems Command, under contract with Bolt, Beranek and Newman, Inc., of Cambridge, Massachusetts [25]. The SCI measurement is patterned after the AI using a 9-band signal-to-noise ratio measurement over the speech spectrum. The data are analyzed in an analog mode with an instrument that computes the signal-to-noise ratio in each band, sums

and normalizes the results, and provides a digital readout display as a decimal value between zero and one. The SCI measure can be written for the 9-band method as

$$SCI = \frac{1}{270} \sum_{i=1}^9 [10 \log s_i/n_i + 12] \text{ dB} \quad (4)$$

where s_i is signal power and n_i is noise power. As with VIAS, SCI produces a measure that is highly correlated with AS but both measures require special equipment and are time-consuming to apply. Therefore, they are not considered acceptable for routine testing of ACSB radios.

Measurements on the ACSB radios indicate that SINAD is very dependent on the audio component in the rf signal; therefore, an approach that makes use of SINAD with some specification of the input audio level is necessary. Using the general test equipment configuration shown in figure 13, SINAD vs. the power in the audio portion of the rf signal was measured for the two ACSB radios. The results are given in figures 14 and 15 for ACSB radios from manufacturers A and B, respectively. SINAD is dependent on the level of the audio component of the rf signal, and in one case, also on the level of the pilot component. This is taken into account by making the audio-to-pilot ratio a parameter.

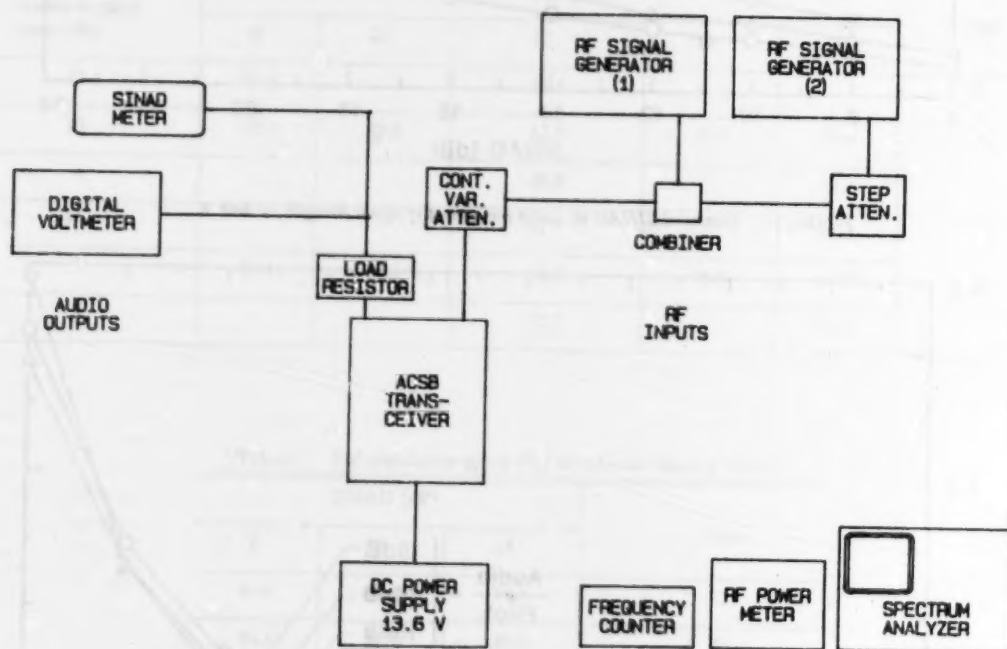


FIGURE 13. General equipment configuration for making ACSB receiver measurements.

6.2 Subjective Measurements

It is easy to design a subjective measure for the purpose of specifying the performance of a voice channel. However, the paucity of designs that include detailed procedures is often a tremendous obstacle to the understanding and acceptance of voice performance measures. For purposes of this work, a subjective measure must be selected that can be correlated with some objective measure. This restricts the subjective tests to those for which both accuracy and repeatability have been exhibited.

The types of subjective tests that have been developed are too numerous to be covered here. The reader is referred to Nesenbergs et al. [26] and to Hoffmeyer [27] for a thorough review of the subject. The two types of testing that have gained credibility are first, the intelligibility tests, and a distant second, the diagnostic tests such as the diagnostic rhyme test [28].

Since the goal is to determine the acceptability of equipment performance and since the diagnostic tests are expensive, they were eliminated from further consideration. Even though some of the diagnostic tests

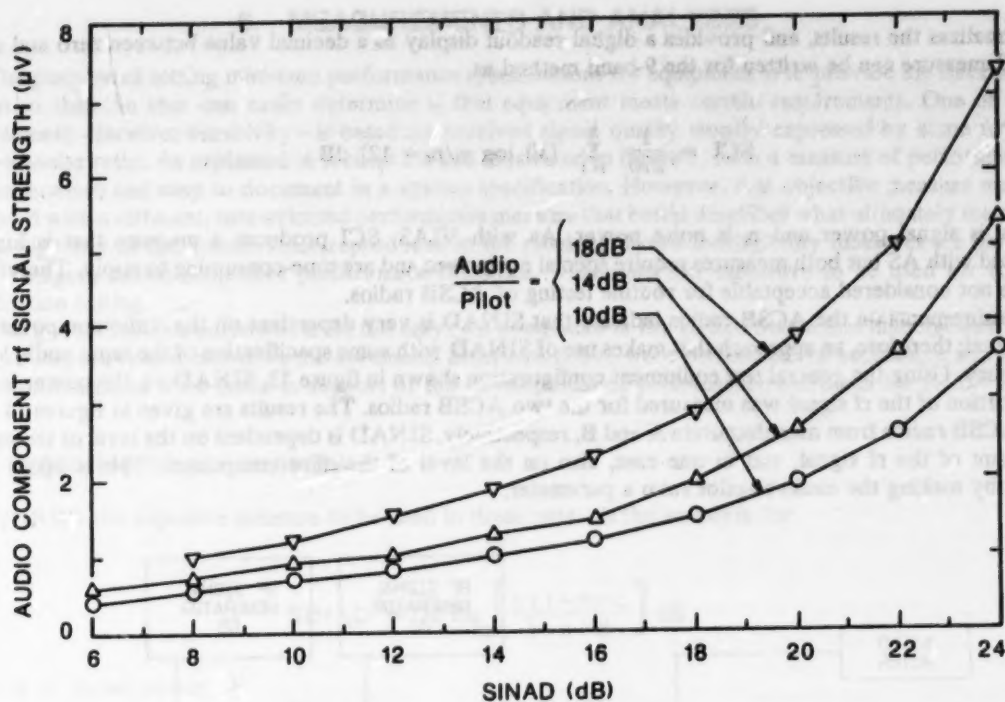


FIGURE 14. Measured SINAD vs. audio component of signal strength for unit A.

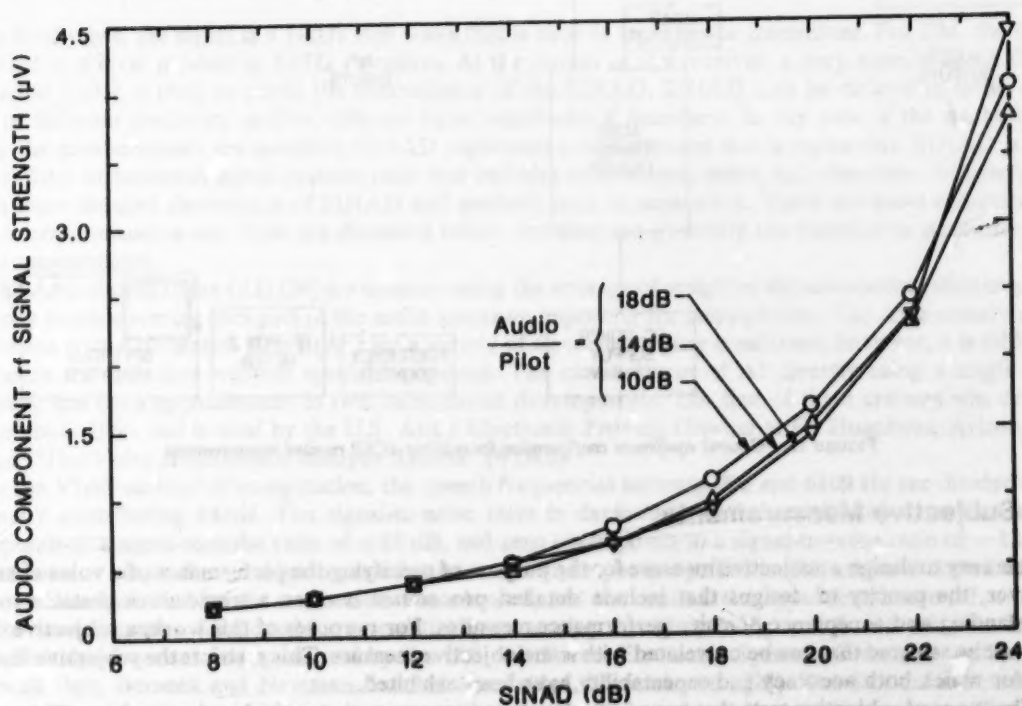


FIGURE 15. Measured SINAD vs. audio component of signal strength for unit B.

measure characteristics other than intelligibility, there is little evidence that these measures are useful for predicting user acceptability for systems that use analog modulation. Thus, the relatively simple intelligibility testing using trained listener panels, called the AS, was used.

The AS uses phonetically balanced word lists in groups of 50 words [29]. These word groups are recorded using several male and female talkers so that one 50-word group represents a cross section of the talkers. The talkers are carefully chosen, usually from radio broadcasters, and the recordings are done under studio conditions with the level on the tape carefully controlled. Appendix B provides more detailed information on the AS and the procedures for making the audio tapes to be scored. The articulation scoring was done by the U.S. Army Electronic Proving Ground, Ft. Huachuca, Arizona, which maintains trained listener panels.

Using the results for SINAD (figs. 14 and 15) as a guide, audio tapes were prepared for some of the combinations of SINAD values and audio-to-pilot ratios. The combinations chosen and the articulation scores obtained are given in table 1. In addition, to support the comparison of ACSB and FM, articulation scores were obtained for two different FM radios operated at 6, 12, and 18 dB SINAD. The articulation scores for the FM radios are given in table 2 and the results for ACSB and FM are compared in section 7.

TABLE 1. ACSB articulation scores (%) for selected values of SINAD and audio-to-pilot ratio

Audio-to-pilot ratio (dB)	SINAD (dB)					Unit
	8	10	12	14	16	
14	86.6		88.4			A
10	76.8	83.9	85.9	89.7	93.4	
6			89.5		91.4	
14			91.7		98.2	B
10	80.4	87.6	91.0	95.0	97.7	
6			91.1		97.9	

TABLE 2. FM articulation scores (%) for selected values of SINAD

SINAD (dB)			Unit
6	12	18	
84.8	88.4	94.1	C
66.4	88.8	96.8	D

6.3 Other Measurements

In order to compare different ACSB transmitters or to compare ACSB transmitters with transmitters using other types of modulation, output power must be specified. It has been shown that the appropriate power measure for ACSB is peak envelope power (PEP). Using the general equipment configuration shown in figure 16, the PEP was measured for the two ACSB radios. Using a static 1-kHz tone varied from a low level to levels where the rf amplifiers began to saturate, the results shown in figures 17 and 18 were obtained. Although both units have a rated PEP of 25 W, neither one could be raised to that level using a static input. Unit A could be raised to about 21 W, but unit B could only be raised to about 14 W. This result suggests the possible need for a dynamic test tone for use in transmitter power measurements.

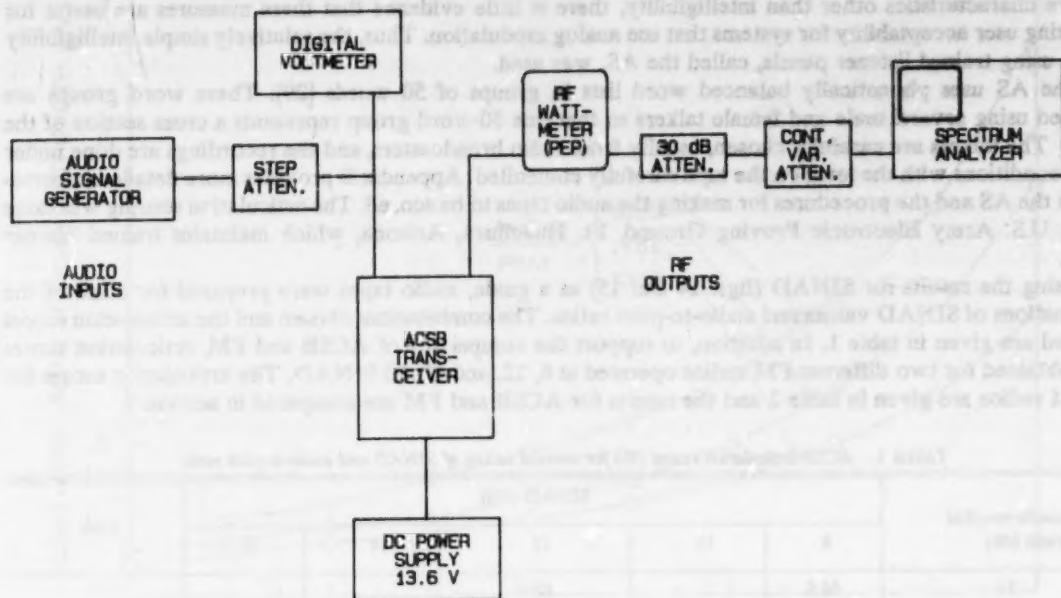


FIGURE 16. General equipment configuration for making ACSB transmitter measurements.

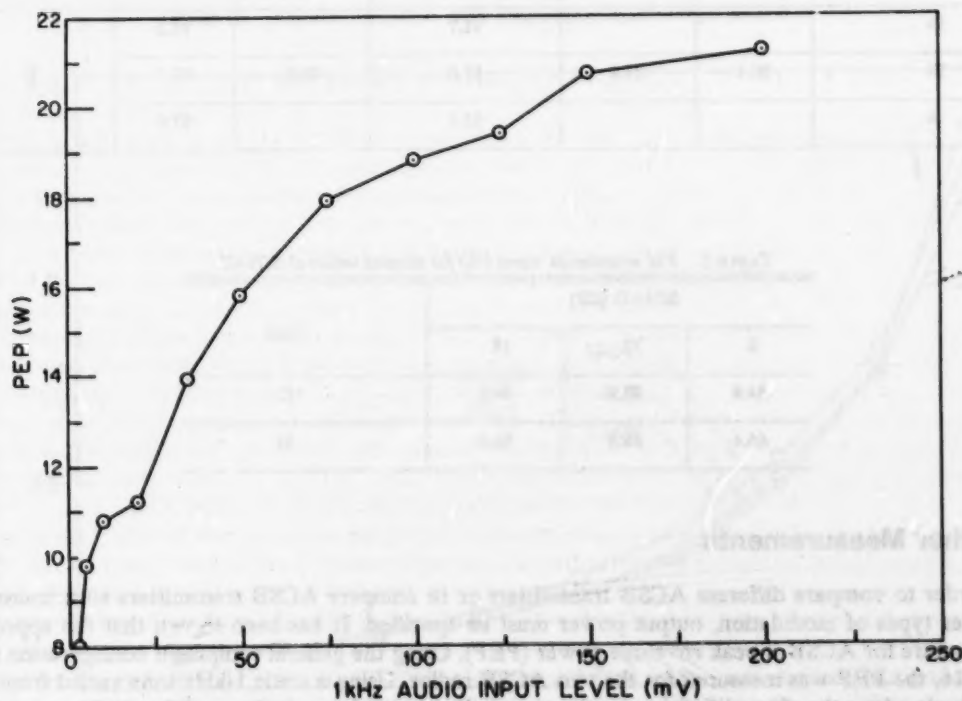


FIGURE 17. Measured peak envelope power vs. audio input level for unit A.

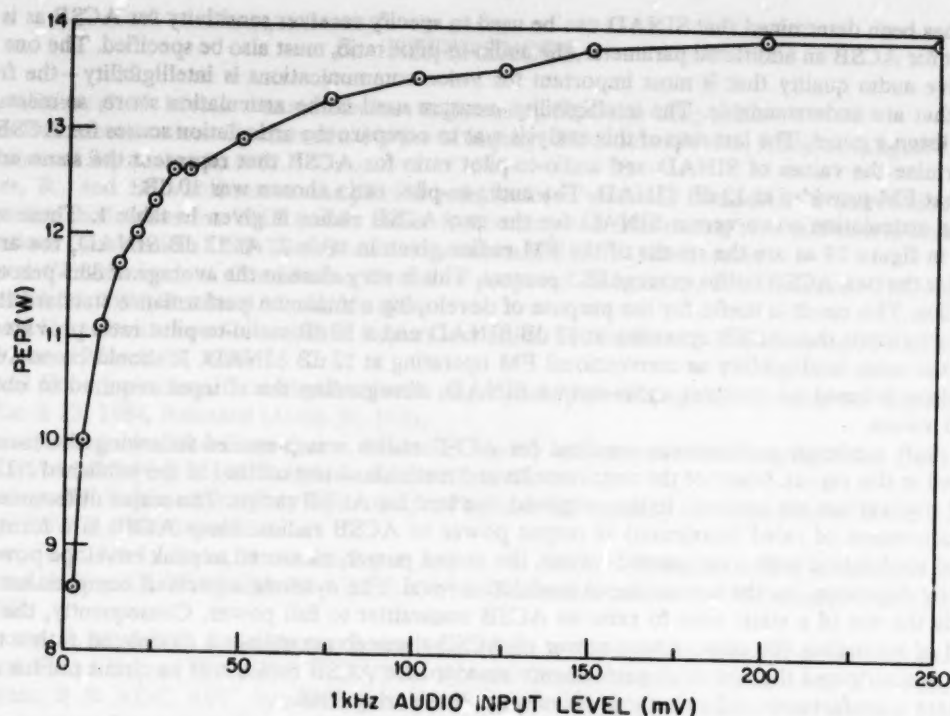


FIGURE 18. Measured peak envelope power vs. audio input level for unit B.

7. SUMMARY AND CONCLUSIONS

Recent technical developments and new FCC rules and regulations have allowed ACSB to be used for land mobile radio communications including law enforcement and public safety applications. ACSB is attractive for land mobile radio applications because it provides immediate relief from overcrowded bands. ACSB allocations provide for a channel width of 5 kHz, which is five or six times less than that required for conventional FM. The spectrum efficiency, partly dependent on adjacent channel protection ratios, is somewhat less—about three to five times better than FM.

ACSB is an implementation of single sideband, suppressed carrier, amplitude modulation with some enhancements to ameliorate the shortcomings of SSB. The singular positive aspect of ACSB is the narrow rf emission. This comes from the fact that SSB modulation has an rf emission bandwidth that is essentially equal to that of the baseband audio signal formed in the audio circuits of the transmitter. The output power of SSB is proportional to the input (speech) power. This shortcoming is partially eliminated by compressing the dynamic range of the audio signal by a factor of four (in decibels). For the typical speech signal with 30 dB of dynamic range, the media output power is increased by 11.25 dB with no change in the peak output power. This power advantage directly benefits the received signal-to-noise ratio at the receiver.

The other major shortcoming of an SSB modulation scheme is the requirement for relatively precise receiver tuning. If the receiver is not tuned to within several tens of Hertz of the correct frequency, audio distortion is evident. A tuning error of 100 Hz would cause significant distortion of the demodulated audio. The ACSB signal includes a pilot tone that is used by a phase locked loop circuit in the receiver to maintain precise tuning. The pilot is also used to control the expanders that restore the audio signal to its original dynamic range. The pilot may also be used for automatic gain control, squelch, and selective signaling.

The "compandor" in amplitude companded sideband is really comprised, then, of two separate circuits: one providing compression in the transmitter and another providing expansion in the receiver.

Does ACSB perform as well as conventional FM? There are two important performance aspects: range (coverage) and audio (speech) quality. It has been shown that ACSB rated using peak envelope power (PEP) and conventional FM rated using RMS power will provide the same range or coverage assuming certain other parameters are equal (i.e., receiver sensitivity, antenna gains, transmission line losses, intervening terrain, etc.). The main purpose of the work presented in this report was to address the question of subjective audio quality. Does ACSB provide the same, better, or poorer audio quality than conventional FM?

It has been determined that SINAD can be used to specify receiver sensitivity for ACSB as is done for FM, but for ACSB an additional parameter, the audio-to-pilot ratio, must also be specified. The one aspect of subjective audio quality that is most important for voice communications is intelligibility—the fraction of words that are understandable. The intelligibility measure used is the articulation score, as measured by a trained listener panel. The last step of this analysis was to compare the articulation scores for ACSB and FM to determine the values of SINAD and audio-to-pilot ratio for ACSB that represent the same articulation score that FM provides at 12 dB SINAD. The audio-to-pilot ratio chosen was 10 dB.

The articulation score versus SINAD for the two ACSB radios is given in table 1. These values are plotted in figure 19 as are the results of the FM radios given in table 2. At 12 dB SINAD, the articulation scores for the two ACSB radios average 88.5 percent. This is very close to the average of 88.6 percent for the FM radios. This result is useful for the purpose of developing a minimum performance standard for ACSB. The conclusion is that ACSB operating at 12 dB SINAD and a 10 dB audio-to-pilot ratio provides approximately the same intelligibility as conventional FM operating at 12 dB SINAD. It should be noted that this comparison is based on receiver audio output SINAD, disregarding the rf input required to obtain these SINAD values.

A draft minimum performance standard for ACSB radios was prepared following the investigations described in this report. Many of the requirements and methods of test utilized in the published NIJ standard for FM transceivers are common to the proposed standard for ACSB radios. The major difference concerns the measurement of rated (maximum) rf output power of ACSB radios. Since ACSB is a form of single sideband modulation with a suppressed carrier, the output power, measured as peak envelope power (PEP), is directly dependent on the instantaneous modulation level. The dynamic aspects of compression appear to preclude the use of a static tone to raise an ACSB transmitter to full power. Consequently, the proposed method of measuring the rated output power of ACSB transceivers utilizes a modulated rather than static tone. It is anticipated that the draft performance standard for ACSB radios will be circulated for review by equipment manufacturers and communications specialists during 1988.

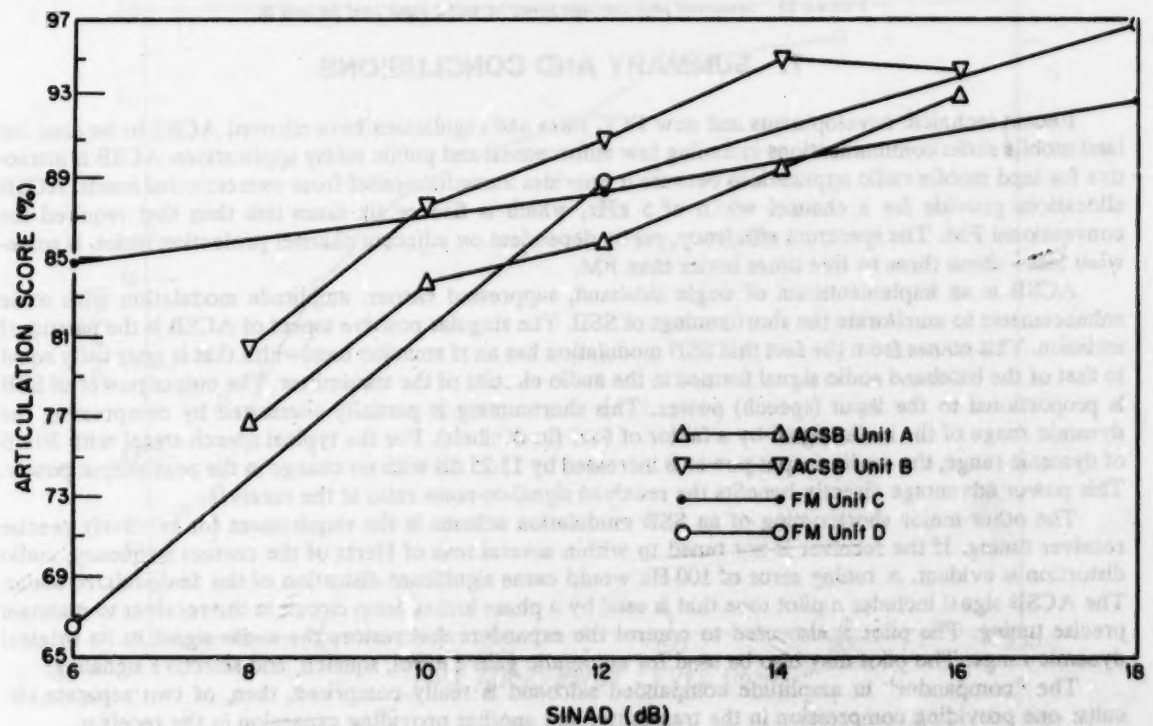


FIGURE 19. Articulation score vs. SINAD for ACSB operating at 10 dB audio-to-pilot ratio compared to FM.

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APPENDIX A. SINAD DEFINITION AND MEASUREMENT METHODS

Before 12 dB SINAD was introduced, 20 dB quieting was used as a reference to measure FM receiver sensitivity. This method involved measuring the audio output of the receiver with no input rf signal and then applying a carrier signal to the antenna input until the audio output was reduced by 20 dB (hence 20 dB quieting). An obvious drawback of this method is that it provides little or no indication of the receiver's true sensitivity—its ability to faithfully reproduce the modulation of a weak signal. SINAD provides a measure (albeit an incomplete one) of the ability of the receiver to accurately reproduce the modulation on the carrier.

SINAD is a measure (at the audio output of the receiver) of the following voltage ratio, expressed in decibels

$$\text{SINAD} = 20 \log \frac{\text{Signal} + \text{Interference} + \text{Noise} + \text{Distortion}}{\text{Interference} + \text{Noise} + \text{Distortion}} \text{ dB.}$$

It should be noted that audio distortion of the signal (regardless of where it is introduced in the transmitter-receiver system) will affect the measurement of SINAD. The audio amplifiers in most land mobile FM receivers can be expected to introduce less than 10 percent distortion. A receiver having an audio amplifier with 10 percent distortion cannot have a SINAD greater than 20 dB. The reference level for receiver sensitivity receiver must be set at a level that reflects the sensitivity of the amplifiers and the quality of intervening stages, not only the fidelity of the audio amplifier. Using 12 dB SINAD as the reference level allows all relevant portions of the receiver to influence the measure of receiver sensitivity.

Several organizations have developed performance standards for testing land mobile receivers. One of these is the National Institute of Justice (NIJ), which has produced NIJ Standard-0209.01, Personal FM Transceivers [1]¹ and NIJ Standard-0210.00, Mobile FM Transceivers [2]. These two documents each include a section on the characteristics of the test equipment needed to perform the 12 dB SINAD receiver sensitivity measurements. Another organization that provides this information is the Electronic Industries Association, whose Standard RS-204-C (EIA RS-204-C) [3] details definitions and methods of measurement of the characteristics of FM and PM land mobile receivers in fixed or vehicular installations. EIA RS-204-C (par. 6.1) defines the minimum specifications for instruments used to measure SINAD. In this area, EIA RS-204-C apparently differs from the previous standard, Electronic Industries Association Standard RS-204-B [4] in two important ways. EIA RS-204-C specifies that the notch filter "shall not cause more than 1 dB of relative attenuation of the total noise output power of a 300 to 3000 Hz bandwidth flat noise source." A more narrow notch filter will reduce the amount of noise that is filtered out by the notch filter. Therefore, for the same signal strength, one might get a lower SINAD reading using an instrument complying with EIA RS-204-C.

EIA RS-204-C also specifies that the voltmeter in the distortion meter must be a true rms voltmeter. EIA RS-204-B specified an average responding voltmeter. The amount of error introduced by using an average responding voltmeter depends on the type of noise and distortion on the signal being measured. The Operating and Service Manual for the HP 339A states: "Third harmonic usually causes the greatest amount of error. For example, when measuring a signal with third harmonic content, an average responding meter can be in error by +5 percent to -20 percent depending on the amplitude and phase of the harmonic relative to the fundamental frequency. Due to errors inherent in average responding meters, a distortion analyzer which employs this type of detector will also be subject to the same measurement errors. These errors can cause indicated distortion readings to be as much as 1.3 dB below the actual rms value for certain combinations of second and third harmonics."

Indeed, preliminary findings seem to indicate that one may obtain a different value when measuring SINAD depending on whether one uses equipment conforming to EIA RS-204-B (par. 6.1) or equipment conforming to EIA RS-204-C (par. 6.1). The indicated SINAD reading was compared for three different instruments:

1. A distortion analyzer that uses a true rms voltmeter and has a very narrow notch filter (thereby conforming to EIA RS-204-C)
2. A SINAD meter conforming to EIA RS-204-B
3. A SINAD meter conforming to EIA RS-204-C

Initial measurements indicate a different SINAD value may be obtained depending on the instrument used.

¹ Numbers in brackets refer to the references at the end of this appendix.

Figure A1 shows the SINAD readings obtained with a distortion analyzer conforming to EIA RS-204-C, a SINAD meter conforming to EIA RS-204-B, and a SINAD meter conforming to EIA RS-204-C. The measurement obtained on the distortion analyzer was used as the reference for each value of SINAD. Eight measurements were made at each SINAD value and the averages of these measurements are plotted on the figure. Averages of eight measurements were used in an attempt to reduce random error and more accurately show systematic error. Readings may vary from reading to reading due to the difficulty of making a reading due to meter flicker as well as other variations in repeatability of all three instruments. Individual readings of the SINAD meters may have varied from the reading of the distortion analyzer more or less than those shown by the averages. The X-axis is the SINAD reading of the distortion analyzer more or less than those shown by the averages. The SINAD indicated by the distortion analyzer, which is obviously a straight line, is plotted as well. The average reading of the SINAD meter conforming to EIA RS-204-B seems to be about 2 dB above the reading on the distortion analyzer. Since the stated accuracy of all three instruments is only 1 dB both the distortion analyzer and the SINAD meter conforming to EIA RS-204-B could be within tolerance over much of the plot. However, the average reading of the SINAD meter conforming to EIA RS-204-C follows the reading on the distortion analyzer conforming to EIA RS-204-C much more closely. It would therefore appear that a somewhat different SINAD reading may be obtained depending on whether the SINAD meter used conforms to the current standard (EIA RS-204-C) or to the older standard (EIA RS-204-B).

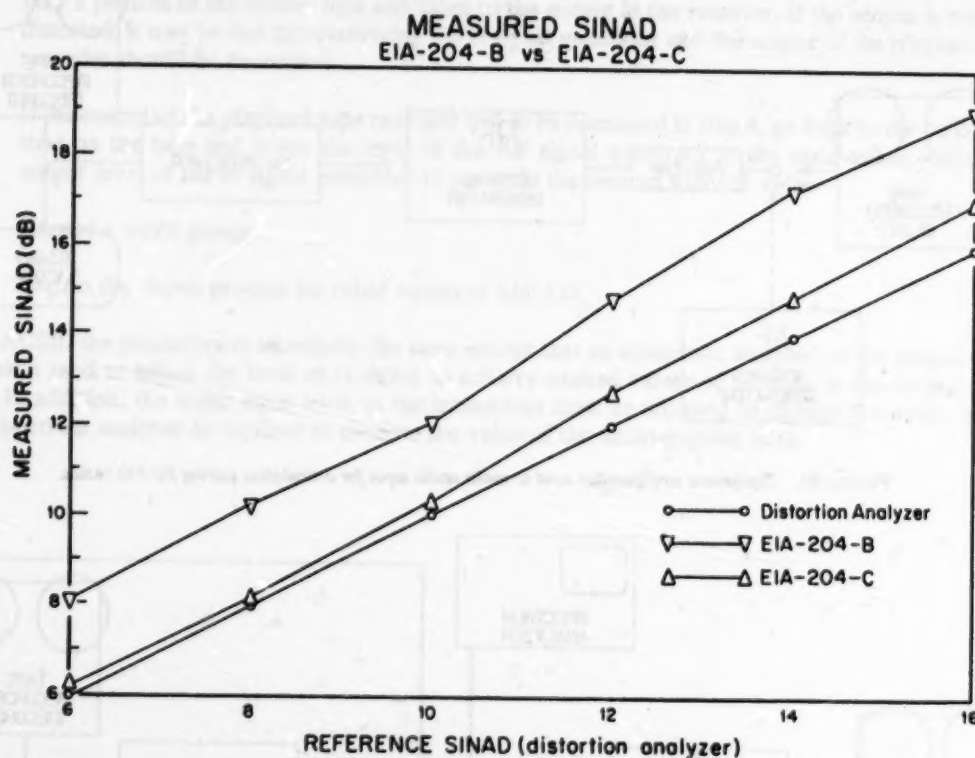


FIGURE A1. Comparison of SINAD measurements made using two SINAD meters—one that conforms to EIA RS-204-B and another that conforms to EIA RS-204-C.

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APPENDIX B. ARTICULATION SCORE MEASUREMENTS

As discussed in this report, the articulation score (AS) is used to represent the subjective measure of system performance. This appendix describes the procedure for making the audio tapes that are then scored by trained listener panels. For this work, the U.S. Army Electronic Proving Ground (USAEPG) at Ft. Huachuca, AZ, was contracted to do the scoring. The trained listener panels are actually maintained by Bell Textron, Inc., in Tucson, AZ, under contract to the USAEPG.

Once the administrative arrangements have been made, USAEPG provides "master" tapes with a number of recorded word groups. Each word group is preceded by a 1-kHz tone used to set playback and record levels. Two equipment configurations are given. Figure B1 shows the configuration for making audio tapes for an FM radio. Here, an rf generator modulated with the playback tape recorder is used to generate the FM signal. Figure B2 shows the equipment configuration for making audio tapes for the ACSB radios. Two ACSB transceivers are used and the one being used as a receiver is placed in a shielded room to isolate it from any undesired emissions. In either case, two high-fidelity, reel-to-reel tape recorders are required as well as the equipment for measuring SINAD and the different level settings.

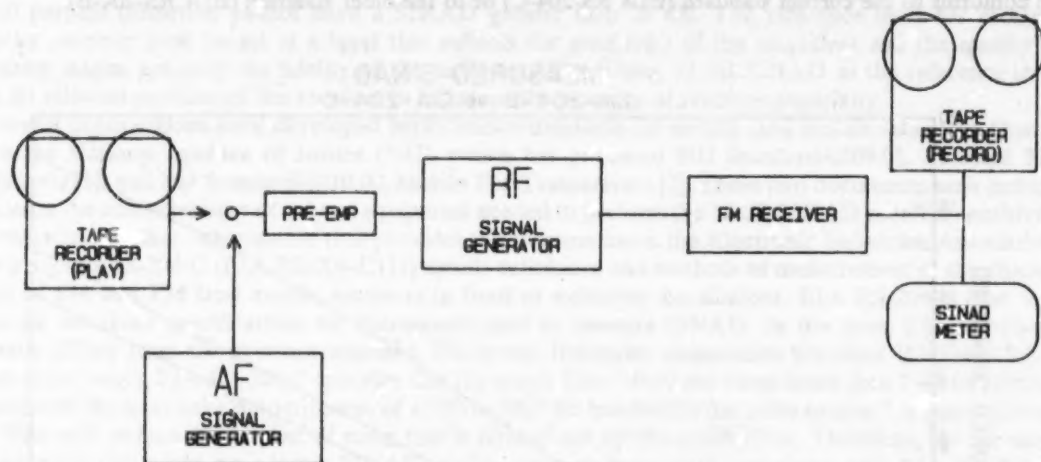


FIGURE B1. Equipment configuration used to make audio tapes for articulation scoring for FM radios.

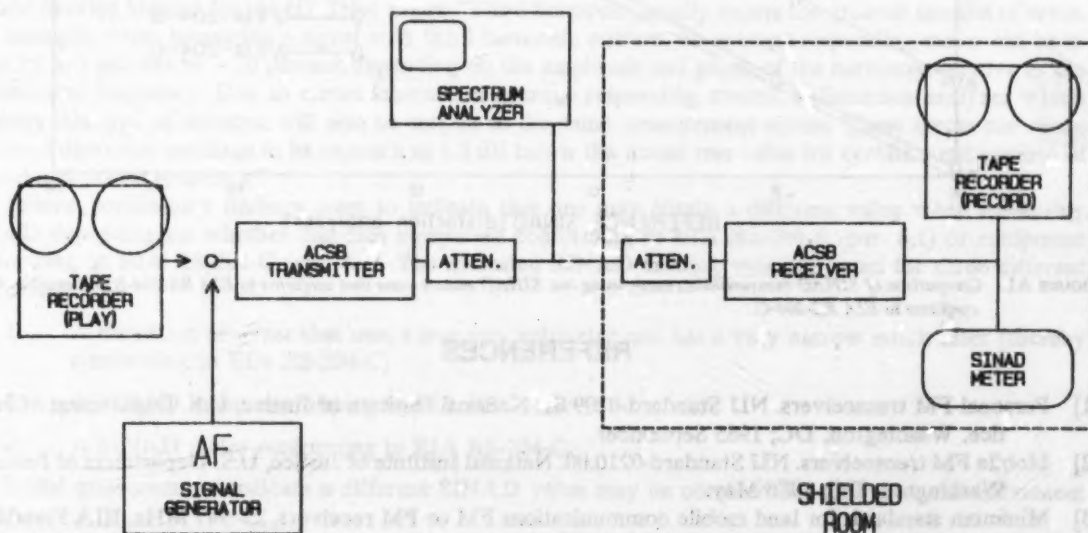


FIGURE B2. Equipment configuration used to make audio tapes for articulation scoring for ACSB radios.

The playback tape recorder channel is routed to both the transmitter and the second (right) channel of the recording tape recorder. Using the calibration tone on a master tape, the levels of the channels should be set to the same level. In addition, the levels should not exceed 0 dB by more than 1 to 2 dB during the time the voice is being played. These levels must not be varied throughout the remainder of the test. The left channel of the recording tape recorder is used for recording the receiver output. It is this channel of the recorded audio tape that is scored.

The general procedure for FM is as follows:

1. Set the signal generator to 1 kHz at a level producing 3-kHz deviation in the rf signal generator.
2. Adjust the output level of the rf signal generator so that a desired value of SINAD is obtained (it is best to start with a large value such as 18 dB).
3. Using a master tape, set the level of the tape recorder output during the 1-kHz calibration tone to the same level as the oscillator output (if possible, ensure that the SINAD reading is unchanged).
4. Play a portion of the master tape and listen to the output at the receiver. If the output is severely distorted, it may be due to overdriving the rf signal generator and the output of the playback tape recorder should be decreased.
5. If the output of the playback tape recorder had to be decreased in step 4, go back to the calibration tone on the tape and lower the level of the AF signal generator to the same value. Adjust the output level of the rf signal generator to maintain the desired SINAD value.
6. Record a word group.
7. Repeat the above process for other values of SINAD.

For ACSB, the procedure is essentially the same except that an attenuator attached to the output of the transmitter is used to adjust the level of rf signal to achieve desired values of SINAD at the output of the receiver. In addition, the audio input level to the transmitter must be adjusted to change the audio-to-pilot ratio. A spectrum analyzer is required to observe the value of the audio-to-pilot ratio.

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